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ABSTRACT

PERFORMANCE ANALYSIS AND RESOURCE ALLOCATION IN CDMA WIRELESS NETWORKS FOR MULTIMEDIA SERVICES

by
Sebnem Zorlu

The proliferation of Internet and demand for wireless services necessitate large increases in capacity and data rates in order to support different multimedia services. Wireless systems will be required to support sources with a variety of traffic characteristics and quality of service requirements. This requires algorithms for admission control and resource allocation at the session, burst and packet levels. The purpose of this research is to develop and analyze optimal resource allocation strategies to maximize throughput of wireless systems with integrated services. Given the multimedia user requirements, the problem addressed can be formulated as a constrained optimization problem. The objective of the admission control and resource allocation policy is to determine the channel allocation to the users with the corresponding feasible power and rate vectors such that overall traffic carried by the system is maximized when all quality of service constraints are met.

The thesis is focused on Code Division Multiple Access (CDMA) wireless networks where transmission modes of the users are controlled according to their QoS requirements and traffic characteristics. The problem under consideration is to find an optimal allocation of traffic channels for a CDMA system with integrated services in order to increase the capacity. Specifically, different channel allocation techniques are examined for data applications, namely circuit, dedicated burst/packet transmission modes and common packet channel transmission schemes. The performance analysis of CDMA common packet channel transmission schemes is studied in depth for finite population and finite buffers/finite sojourn time cases for more realistic data

arrival processes than assumed in the literature. The effect of model parameters and user characteristics such as packet length distribution on the system behavior is quantified.

PERFORMANCE ANALYSIS AND RESOURCE ALLOCATION IN CDMA WIRELESS NETWORKS FOR MULTIMEDIA SERVICES

by
Sebnem Zorlu

A Dissertation
Submitted to the Faculty of
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Doctor of Philosophy in Electrical Engineering

Department of Electrical and Computer Engineering

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APPROVAL PAGE

PERFORMANCE ANALYSIS AND RESOURCE ALLOCATION IN CDMA WIRELESS NETWORKS FOR MULTIMEDIA SERVICES

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For all the wonderful things he has done for me, this thesis is dedicated to my uncle
TAMER TEKMAN

"Dünyada kiracı gibi değil
Yazlığına gelmiş gibi de değil,
Yaşayın dünyada babanızın eviymiş gibi
Tohuma, toprağa, denize inanın,
İnsana hepsinden önce
Bulutu, makineyi, kitabı sevin
İnsanı hepsinden önce.
Kuruyan dalın
Sönen yıldızın
Sakat hayvanın
Duyun kederini
Ama hepsinden önce de insanın
Sevindirsin sizi cümlesi nimetlerin
Sevindirsin sizi karanlık ve aydınlık,
Sevindirsin sizi dört mevsim,
Ama hepsinden önce insan sevindirsin sizi"

Nazım Hikmet (1902-1963)

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CHAPTER 1

INTRODUCTION

The proliferation of Internet and demand for wireless services necessitate large increases in capacity and data rates in order to support different multimedia services. This market force is also driving standards development process. The radio transmission technology of the third-generation mobile communication system should be able to support multimedia, personal as well as intelligent functions. Specifically, the system should meet the requirements recommended by International Telecommunications Union-Radiocommunication Sector (ITU-R) such as advanced multi-rate services up to 2Mbps and a quality comparable to that of the fixed communication networks. At the same time, the system should aim to realize a simple cell structure, easy channel management, high subscriber capacity and low transmit power. Wireless systems will be required to support sources with a variety of traffic characteristics and quality of service (QoS) requirements. This requires algorithms for admission control and resource allocation at the session, burst and packet levels.

The purpose of this research is to develop and analyze optimal resource allocation strategies to maximize throughput of wireless systems with integrated services. Given the multimedia user requirements, the problem addressed can be formulated as a constrained optimization problem defined in the following section.

1.1 Problem Statement

Let N be the number of users in the system. Each user has quality of service, power and rate requirements. Define the power and rate allocation vectors for the N users to be $\mathbf{P} = [P_1, P_2, ..., P_N]$ and $\mathbf{R} = [R_1, R_2, ..., R_N]$ respectively. Power and rate allocation vectors are said to be feasible if they meet the user requirements, i.e., $0 \leq P_i \leq P_i^{max}$ and $R_i^{min} \leq R_i \; \forall i = 1...N$ respectively where P^{max} and R^{min} correspond to peak instantaneous values. The tolerable QoS for each user is in the forms of maximum bit error rate (BER) or frame error rate (FER) denoted by $\mathbf{\Gamma} = [\Gamma_1, \Gamma_2, ..., \Gamma_N]$ and maximum delay denoted by $\mathbf{D} = [D_1, D_2, ...D_N]$. The objective of the admission control and resource allocation policy is to determine the channel allocation to the users with the corresponding feasible power and rate vectors such that overall traffic carried by the system is maximized when all QoS constraints are met with equality.

1.2 Research Objective

The thesis is focused on Code Division Multiple Access (CDMA) wireless networks with multimedia services where the transmission modes of the mobiles are controlled according to their QoS requirements and traffic characteristics. Specifically, two main traffic classes are considered: real time services and non-real time services. The first class consists of delay intolerant applications requiring a constant bit rate of R_i^1 bits/s and a bit error rate (BER) of at most Γ_i^1, while the second class covers delay tolerant applications requiring a bit rate of at least $R_{i_{min}}^2$ bits/s and a bit error rate of at most Γ_i^2. The addressed problem is to find an optimal allocation of traffic channels for a CDMA system with integrated services in order to increase the capacity.

According to the allocation control at session, burst and packet level, the transmission modes are distinguished as dedicated circuit, dedicated burst/packet and common channel packet switching. In circuit switching, the users are allocated a dedicated channel and a continuous connection is guaranteed during a session while in packet switching channel allocation is done for a packet duration. Circuit and packet switching techniques are insufficient in meeting the QoS requirements of bursty long multimedia messages due to the poor channel utilization and high per packet delay respectively. The proposed burst switching technique in cdma2000 MAC layer ([42]) aims to overcome these problems by allocating the dedicated channels to the burst of data and releasing them at the end of the bursts. Furthermore, this scheme enables the utilization of one code among multiple users while still meeting their QoS requirements. The advantage of burst reservation schemes for data services is the minimization of the interference for voice and data packets at the expense of higher overhead to control and measure the channel load (higher than circuit mode and common channel transmission modes). Aloha type common packet transmission requires higher rate of retransmission for data users while a simpler control mechanism is needed [22].

1.3 CDMA Wireless Networks

Code Division Multiple Access (CDMA) has received a great deal of attention as a multiple access method for future mobile networks. Its main advantages are higher radio capacity and capability of flexible data transmission. CDMA is a key technology in the submitted proposals to the ITU on Universal Mobile Telecommunications System/International Mobile Telecommunications in the year 2000 (UMTS/IMT-2000). The Association for Radio Industry and Business (ARIB) in Japan became the driving force behind a third generation radio transmission technology known as wideband CDMA (WCDMA). The European Telecommunications Standards Institute (ETSI) and ARIB have managed to merge their technical proposal into one harmonized WCDMA standard air interface. In the United States, The Telecommunications Industry Association (TIA) has proposed two air interface standards. One of them is CDMA based interface, referred to as cdma2000, that maintains backward compatibility with existing IS-95 networks.

South Korean Telecommunications Technology Association (TTA) supports two air interface proposals, one similar to WCDMA and the other to cdma2000.

Efficient bandwidth utilization is crucial in providing multi-media services on a cellular CDMA platform. In a CDMA network, the controllable resources of the users are their powers and data rates. Therefore, in order to support effective data service in a CDMA wireless environment with limited resources, the effective use of these resources in wireless medium access control/radio link control (MAC/RLC) protocols should be considered.

The problem of jointly controlling the date rates and transmit powers of users in cellular Direct Sequence CDMA (DS-CDMA) networks can be solved by using Multi-Code CDMA (MC-CDMA) or Variable-Gain DS-CDMA (VG-CDMA). In MC-CDMA, the processing gain of each code is fixed and multiple codes are assigned to the high rate users. In VG-CDMA, all users use a single code with varying processing gain [1].

CHAPTER 2
PREVIOUS WORK

This section provides a brief overview of the previous work reported in the literature. Since the thesis spans many different aspects related to the resource allocation problem in a CDMA system supporting integrating services, the review of the previous work is divided into three main parts: capacity of wireless CDMA networks for dedicated circuit and burst/packet transmission modes, throughput of common channel CDMA systems and resource allocation techniques in wireless CDMA networks.

2.1 Capacity of Wireless CDMA Networks

Early works of Lee [2], Gilhousen et al. [3] and Wilson et al. [4] on capacity analysis of code division multiple access (CDMA) cellular systems shows that power-controlled CDMA systems promise a large increase in cellular capacity when compared to other multiple access techniques. In Viterbi et al. [5], the reverse link capacity of a CDMA cellular voice system is evaluated. Erlang capacity is computed for a system with power control and variable rate vocoder based on voice activity. The uplink CDMA capacity is shown to be limited by the outage probability defined when the total interference (I_0) of the users within the same cell and other cells exceeds the background noise level (N_0) by an amount $1/\eta$.

If the number of users in a class is a Poisson random variable with mean λ/μ ($P(k) = \frac{(\lambda/\mu)^k}{k!} e^{-\lambda/\mu}$) and each user has an activity factor and requires different energy to interference ratio (E_b/I_0), the outage condition is

$$\sum_{i=1}^{k} \nu_i E_{bi} R + \sum_{j}^{other\ cells} \sum_{i=1}^{k} \nu_i^j E_{bi(j)} R + N_0 W \leq I_0 W \qquad (2.1)$$

where k represents the number of users and is a Poisson random variable with mean λ/μ, ν is a binary random variable taking values 0 with probability $(1-\xi)$ and 1 with probability ξ, R is the data rate and W is the spread spectrum bandwidth. The nonblocking condition in this case can be written as:

$$Z = \sum_{i}^{k} \nu_i E_{bi}/I_0 + \sum_{j}^{other\ cells} \sum_{i=1}^{k} \nu_i^j E_{bi(j)}/I_0 \leq W/R(1-\eta) \tag{2.2}$$

For a perfect power control system, a constant number of N_u users with the same required bit energy per interference ratio (E_b/I_0) can be supported if:

$$\begin{aligned} N_u E_b R(1+f) + N_0 W &\leq I_0 W \\ N_u &\leq \frac{W/R}{E_b/I_0} \frac{1-\eta}{1+f} \end{aligned} \tag{2.3}$$

where f is the ratio of other cell interference. More detailed computation of other cell interference can be found in Lee [2], Gilhousen et al. [3] and Ayyagari et al. [6]. The latest reference covers also the capacity analysis for non-uniformly loaded cells.

The Erlang capacity of a CDMA system can then be computed setting the blocking probability ($P_b = Pr[Z > (W/R)(1-\eta)]$) to a given threshold. The typical values for voice communications is $\eta = 0.1$, $P_b = 0.01$. Z is a random variable that depends on the random variables $\nu, k, E_b/I_0$. E_b/I_0 ratio of a single user depends on the inaccuracy in power control mechanism and is shown to be approximately lognormally distributed ($P(E_b/I_0) = 10^{x/10}, x \sim N(m, \sigma)$) with a standard deviation between 1 and 2dB for voice communications under closed-loop power control. In environments with excessive multipath a lognormal probability density with mean and standard deviation 7 and 2.4 dB respectively is used. For perfect power control and Poisson random variable k, Z is Poisson random variable with mean $\xi\lambda/\mu$. If k is constant, then Z is binomial random variable with mean $k\xi$. For imperfect power control, the bounds on the outage probability can be found under Gaussian or Lognormal approximations Sampath et al. [7].

The capacity computation has been extended for an integrated voice and data CDMA system. For a CDMA system with circuit-mode transmission the bit energy to noise ratio in a single cell Sampath et al. [8], Ramakrishna et al. [9] is

$$(\frac{E_b}{N_o})_i = \frac{W}{R_i}\frac{P_{r_i}}{\sum_{j\neq i} P_{r_j} + \eta} \tag{2.4}$$

where P_r is the received power at the base station. Here, the activity factor and other cells interference are omitted.

For a given set of classes, at the optimal solution all QoS constraints are met with equality and the optimal power vector is one that achieves all rate constraints with equality. Therefore, the optimal power vector for N users satisfies:

$$\frac{W}{R_i}\frac{p_{l_i} P_i}{\sum_{j\neq i} p_{l_j} P_j + \eta} = \Gamma_i \ \forall i = 1, ...N \tag{2.5}$$

where p_l is the path loss between the mobile and the base station. The optimal received power levels are then computed as:

$$\begin{aligned} P_{r_1} &= \frac{\eta}{(1-\sum_{j=1}^{N}\frac{1}{W/(R_j\Gamma_j)+1})(W/(R_1\Gamma_1)+1)} \\ P_{r_i} &= P_{r_1}(W/(R_1\Gamma_1)+1)/(1+W/(R_i\Gamma_i)) \quad i = 2, ..., N \end{aligned} \tag{2.6}$$

Hence, positivity of the power vector implies:

$$\sum_{j=1}^{N}\frac{1}{W/(R_j\Gamma_j+1)} < 1 \tag{2.7}$$

If the power constraints are included, the following inequality is obtained:

$$\sum_{j=1}^{N} \frac{1}{W/(R_j\Gamma_j+1)} \leq 1 - \frac{\eta_0 W}{min_i[P_i h_i (W/(R_i\Gamma_i+1)]} i = 1...N \tag{2.8}$$

If the activity ratio is included the feasibility condition and the required received power can be written as Sampath et al. [7]:

$$\sum_{j=1}^{N} \frac{\nu_j}{W/(R_j\Gamma_j+1)} < 1 \tag{2.9}$$

$$P_{r_1} = \frac{\eta}{(1-\sum_{j=1}^{N}\frac{\nu_j}{W/(R_j\Gamma_j)+1})(W/(R_1\Gamma_1)+1)} \tag{2.10}$$

Note that, activity factor of real time services such as voice calls reflects the on and off times of the user and is defined from the service traffic characteristics while for non-real time services, it denotes the transmit attempts that can be controlled in an integrated system such as burst reservation schemes or that are not controlled and therefore can require retransmissions such as Aloha type systems.

If the outage condition is also imposed, the feasibility of the power control becomes:

$$\begin{gathered}\sum_{j=1}^{N} \nu_j P_j + N_0 W \leq \frac{N_0 W}{\eta} \\ \sum_{j=1}^{N} \frac{\nu_j}{W/(R_j\Gamma_j+1)} < 1-\eta\end{gathered} \tag{2.11}$$

It can be seen that this condition is more strict than Eq. 2.7 and 2.9 .

In Matragi et al. [10], in addition to the outage probability the Erlang capacity is computed using different call block probabilities for voice and data services. The data call model is a circuit model and assumes that once a data call is accepted in the system, data messages are transmitted without delay throughout the duration of the call. For capacity partitioning [10] model where a fixed fraction of the reverse link capacity is dedicated to each type of class, they use the following equality (Eq. 2.7) by neglecting η:

$$\frac{N_v}{W/(R_v\Gamma_v+1)}+\frac{N_d}{W/(R_d\Gamma_d+1)}=1 \tag{2.12}$$

where indices v and d denote voice and data calls respectively. N_v and N_d are the number of active simultaneous voice and data calls. The maximum number of calls (N^{max}) that can be supported by a cell is defined as the number that exceeds admissible active call numbers (N) with probability less than P_0. For the voice calls, this number is computed using the voice activity factor:

$$N_v^{max} = max(I : \text{such that} \sum_{k=N_v}^{I} \binom{I}{k} \xi^k (1-\xi)^{I-k} \leq P_0) \tag{2.13}$$

The same number is found for data users using the data activity factor. This number (N_v^{max} and N_d^{max}) is then used to compute the blocking probability P_b from the classical Erlang-B formula:

$$P_b = \frac{e^{N^{max}}/N^{max}!}{\sum_{k=0}^{N^{max}} e^k/k!} \tag{2.14}$$

For dynamic sharing model, the total number of available resources is determined by the largest N^{max}. If for example, $N_v^{max} \geq N_d^{max}$ then the total number of unit trunks is N_v^{max} and voice calls consume one unit trunk while each data call consumes N_v^{max}/N_d^{max} unit trunks. The Erlang capacity in this case is found by using the generalized Erlang model Kaufman et al. [11].

In Capone et al. [12], the authors analyze a protocol that admits data traffic into the CDMA cellular system based on the current aggregate voice interference level in order not to degrade the quality of service for the delay critical voice traffic. An on-off voice traffic model and a Poisson data traffic model are considered where no new arrivals are generated by the busy user until the current transmission is complete. Using the outage condition described above, the data rate under perfect power control is chosen as:

$$R_d = \begin{cases} \frac{W/R_v(1-\eta)-Z}{1/R_v N_d (E_b/I_0)_d} & 0 \leq Z \leq (W/R_v)(1-\eta) \\ 0 & \text{otherwise} \end{cases} \tag{2.15}$$

where $Z = \sum_{i=1}^{N_v} \nu_i (E_b/I_0)_v$ and v and d denote voice and data calls respectively.

For log-normal variation in power control, a scaling coefficient $0 \leq c \leq 1$ is introduced to limit the amount of interference by the addition of data as follows:

$$R_d = \begin{cases} \frac{W/R_v(1-\eta)c-Z}{\overline{G}} & 0 \leq Z \leq (W/R_v)(1-\eta)c \\ 0 & \text{otherwise} \end{cases} \tag{2.16}$$

where $\overline{G} = 1/R_v E[N_d] E[(E_b/I_0)_d]$ with Poisson random variable N_d and lognormally distributed $(E_b/I_0)_d$.

In Sun et al. [13], capacity analysis of a multiservice CDMA system with continuous (circuit mode) and discontinuous (packet mode) transmission schemes is studied. It is assumed that synchronization messages are transmitted through traffic channels. Low transmission bit rates (l_i) are transmitted during idle periods to maintain synchronization. The overhead interference (h) is contributed by synchronizing messages used to reestablish physical link when a discontinuous transmission scheme is adopted. In this case the outage condition is written as the following:

$$Z = \sum_{i}^{k} \nu_i (1+h) r_i E_{bi}/I_0 \leq W/R(1-\eta) \tag{2.17}$$

where r_i represents the multipliers of basic rate R and ν_i is a binary random variable taking values 1 and 0 when the user is active and inactive respectively. Therefore, $P(\nu_i = 1) = \xi$ and $P(\nu_i = l_i) = 1 - \xi$, $l_i = 0$ for the discontinuous service and $h = 0$ for continuous service. The discontinuous transmission is implemented for short packet data users where activity factor is taken as 1 during packet duration and 0 otherwise. It is shown that the hybrid transmission scheme can provide higher capacity. The Erlang system capacity assuming that Z is a Gaussian variable is computed. Voice, long and short packet services are considered where closed loop power control is applied for voice and long packet data. Therefore, the SIR of continuously transmitted data is lognormal random variable while SIR of discontinuously transmitted short packet data is a chi-square random variable with two degrees of freedom (exponential random variable). The call admission scheme proposed in [13] introduces the average number of times that a data message must be transmitted before its successful receipt. More detailed analysis of packet switching CDMA networks with Automatic Repeat Request (ARQ) is reviewed in the next section.

2.2 Throughput of Common Channel CDMA Systems

The analysis of slotted (synchronous) and unslotted (asynchronous) random access spread spectrum packet radio networks has been an active research topic for many years. Since error control via acknowledgments and retransmission in non-real time applications is crucial, especially in the environments where message losses are usually higher, a retransmission scheme is used. The packets received with uncorrectable errors are retransmitted (Figure 2.1).

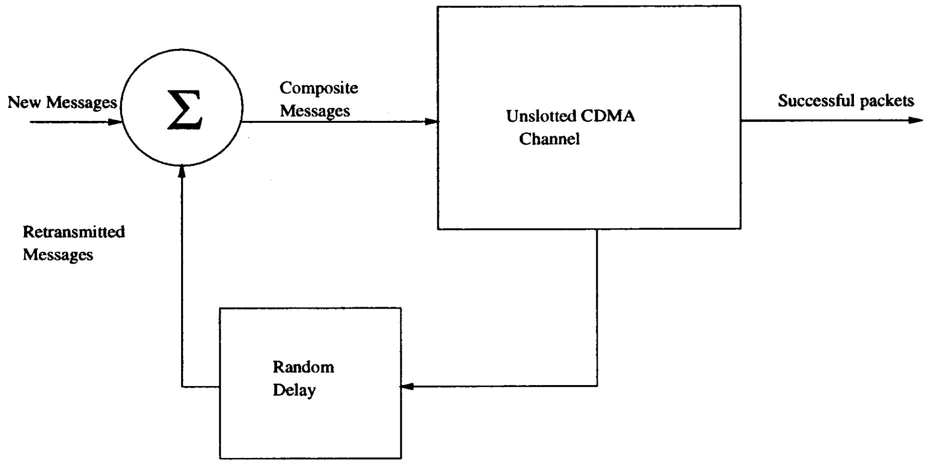


Figure 2.1 Schematic Representation of CDMA Random Access Channel

One of the early papers about slotted random access packet CDMA systems is Raychaudhuri et al. [14] where steady-state throughput characteristics are obtained. The analysis of multicode spread slotted Aloha in Dastangoo et al. [15] is done for code acquisition and packet transmission stages. This work is based on Raychaudhuri et al. [14] and Zhang et al. [16]. Analysis of slotted systems is easier since the number of interferers during a packet transmission is constant while in unslotted case the interfering packets can enter and leave the system during a tagged packet transmission.

In Joseph et al. [17],[18] the stability and performance of the unslotted random access CDMA systems are investigated for a Poisson arrival model with exponential distributed packet length. Their assumption is that any user in backlogged mode is inhibited to generate a new packet. Hence, queuing analysis is omitted. This study has been the reference for many related works. In their analysis, the probability of packet success P_S is defined as:

$$P_S(\tau, \lambda) = \sum_{\overline{m}} P_c(m_1)P_c(m_2)...P_c(m_L)p(m1, m2, ..., m_L|\lambda, \tau) \quad (2.18)$$

where λ is the input traffic rate (new and retransmitted packets), m_i is the number of interferers at the i^{th} bit, L is the number of bits per packet, P_c is bit error rate and p is the joint distribution of m_i's. For a given data rate R, random variable for packet duration is $\tau = L/R$. The multiuser interference process can be represented by a discrete time finite Markov process. Modeling with Markov Process is valid also for an arbitrary distribution if multiuser interference changes slowly relative to a bit duration ($\Delta\tau$ sec).

$$p(m_k|m_{k-1}) = Prob(m_k = j|m_{k-1} = i) = \begin{cases} i\mu\Delta\tau & \text{for } i = j+1 \\ 1 - \lambda\Delta\tau - i\mu\Delta\tau & \text{for } i = j \\ \lambda\Delta & \text{for } i = j-1 \\ 0 & \text{otherwise} \end{cases} \quad (2.19)$$

where $1/\mu = E[\tau]$ is mean packet duration.

Using the Markovian property of the joint distribution of number of interferers, P_S can be written as

$$P_S(\tau, \lambda) = \sum_{\overline{m}} P_c(m_1)P_c(m_2)...P_c(m_L)p(m_L|m_{L-1})...p(m_2|m_1)p(m_1) \quad (2.20)$$

If the composite message length distribution is denoted by $b(\tau)$, then the message flow and packet flow equations can be written as

$$(N - n)\lambda_i = \lambda \int_0^\infty b(\tau)P_S(\tau, \lambda)d\tau$$

$$
\begin{aligned}
S_{in} &= (N-n)\lambda_i A = \lambda \int_0^\infty \tau b(\tau) P_S(\tau, \lambda) d\tau = S_{out} \qquad (2.21)\\
\lambda &= n\lambda_r + (N-n)\lambda_i
\end{aligned}
$$

where N is the number of users in the system, n is the number of backlogged users , λ_i is packet rate per user and $\lambda_r = 1/T_r$ is the retransmission rate and S_{out} is the throughput of the system. Note that stability of the system depends on retransmission delay that must be longer for heavier traffic.

In Yin et al. [19], performance analysis for unslotted CDMA with fixed packet length is investigated for infinite number of users. This work uses L-channel model where an error occurs if the number of interferers exceeds L-1. In Storey et al. [25], a DS-CDMA radio channel with hard decision Viterbi decoding is analyzed for finite population with infinite buffer size. The authors assume heavy traffic condition where they do not distinguish between radios that are backlogged and those that are not. In Okada et al. [26],[27], CDMA slotted/unslotted Aloha systems with finite population and finite buffer size are investigated for fixed packet length.

More general (not Markovian) systems for different retransmission algorithms are investigated via simulation. In Harpantidou et al. [28], slotted random multiple access under bursty packet traffic is studied experimentally; the performance of ideal and implementable slotted Aloha and m-ary stack algorithms are investigated for packet interarrival times distributed according to Pareto distribution. Linear and exponential back-off techniques are investigated in Khan et al. [29] for the uplink common channel packet transmission in Wideband CDMA for Poisson arrival process.

In Soroushnejad et al. [24], the authors present the performance analysis of a CDMA access scheme with integrated services where a slotted Aloha protocol with retransmission control via channel sensing is used by the data users. Each time slot is equal to the duration of single data packet. Voice and data traffic is modeled by Bernoulli process models with fixed packet length where no new packet is generated during backlog mode of data users. An adaptive slotted Aloha DS-CDMA access scheme is proposed in Sallent et al. [22] for voice and data services where data terminals change their transmission rate according to the total (voice+data) channel occupancy to obtain the minimum possible data delay. The analysis is done under the assumption of a Poisson arrival process and exponentially distributed message length for conventional S-Aloha DS-CDMA system and the adaptive scheme is studied via simulation. Fewer articles have analyzed integrated systems for unslotted CDMA systems. In Sandouk et al. [30], outage probability for voice users and throughput for data users are evaluated for an integrated unslotted CDMA system for fixed packet length. In this work, the interference from both media is assumed to be the same and the number of simultaneous transmissions from voice users is assumed to be constant during one data packet. The same assumptions are used in Sato et al. [31] where an integrated voice and data system over a CDMA unslotted Aloha with Channel Load Sensing Protocol is analyzed.

2.3 Resource Allocation in Wireless CDMA Networks

Power control in code division multiple access (CDMA) cellular systems is a crucial issue since the capacity of CDMA networks is mainly interference limited ([2],[3]). Present CDMA cellular systems have been optimized for voice transmission. For voice CDMA systems based on the IS-95 standard, power control is used to combat the near-far problem by maintaining nearly constant received power at the base station. Power control is used as a means of minimizing multi-user interference and improving capacity by adjusting the powers to obtain the same carrier to interference power ratio on all links. In IS-95 reverse link, the signal from each mobile unit should arrive at the base station with the minimum signal-to-noise ratio (SNR) needed to maintain desired quality. In reverse link-open loop control, mobile unit estimates path loss from cell site by comparing received power to transmitted power. Mobile adjusts its power such that the transmitted power is lowered if the signal is determined to be too strong or is increased slightly otherwise. In reverse link-closed loop control, the demodulator at cell site compares received SNR to desired value and commands adjustment accordingly every 1.25 ms. Power is adjusted by a predetermined increment (.5 dB) each time. In forward link, at certain locations, the signal received by a mobile unit may be too weak to accurately decode data due to the excessive shadowing and interference from a neighboring cell. The cell periodically reduces the transmitted power in order not to transmit high power if not necessary. When a mobile detects an increase in its frame error rate, it requests higher power and the cell site increases power by a predetermined amount (.5dB) once per vocoder frame or about 15-20 msec. The dynamic range is limited to +/- 6 dB.

The Carrier-to-Interference Ratio (CIR) (or Signal-to-Interference Ratio (SIR)) balancing technique for power control purposes is presented in several works (Nettleton et al. [32], Zander et al. [33]). The power control algorithms in the literature can be classified as distributed and centralized algorithms where the path gains are known a priori or measured C/I ratio on the active links are utilized. Two distributed power control approaches have been proposed in the literature. In the first approach, the receiver's signal-to-interference ratio (SIR) is measured and the transmission power is adjusted according to whether SIR is below or above some target value. The drawback of this algorithm is that the local adjustments without a global consistency increase the interference to the neighboring areas that result in increase of power in this area, hence an increase of interference. In the second approach, the transmitted power is adjusted in order to balance the SIR's of all links and to maximize the worst SIR in the channel. The drawback of this approach is that during the iterations of power adjustment or after reaching the steady state, the SIR's of the links may fall below the required value. In both approaches, the potential of admitting a new call cannot be foreseen.

In Bambos et al. [34], the optimization problem includes transmitter power minimization, network capacity maximization and optimal resource allocation. The desired properties of a power control scheme are stated as to be distributed (at the node or link level) in order to require minimal usage of network resources for control signaling, simple to be suitable for real-time implementation, agile for fast tracking and adaption to the channel changes and mobility, robust to be able to adapt to stressful contingencies and scalable to perform at various network scales of interest.

In this algorithm, the active links adjust their power as in the standard power control case while the the new ones power up gradually at a constant power rate. When it is not possible to admit any new link, voluntary dropout (VDO) of a new link can mitigate congestion and may help other new links gain admission. For this purpose, SIR increase of new links is used to expect the likelihood of gaining admission soon.

In case of multiple channels, the mobile can switch to the lowest-power channel in a smooth way. In case of multihop wireless network, the minimum power routing is performed. The shortest path from the source to the destination is searched where the arc lengths are the powers needed for establishing the links. Another important issue is the decision of transmitting or waiting according to the channel measurements.

In Ayyagari et al. [35], a power control algorithm is proposed for the problem of admission control in a DS-CDMA network with integrated services. The algorithm is shown to be optimal in the sense of maintaining active link quality (QoS of active users) while maximizing free capacity of new admissions (bandwidth to new users). Bursty packet applications can introduce high interference during active periods. Multiaccess interference is regulated by controlling the transmit powers of the users for active link quality protection. This is done by computing the "interference margin" that is the amount of excess interference that can be tolerated by active users without violating their SNR thresholds. The admission control process is executed in two distinct phases:

1) Scaling: The powers of the active users are scaled to maximize the interference margin. In this optimization procedure, the controllable variable is the power allocation vector for active users. The objective is to maximize the interference margin. The constraints include the SNR requirements of all active links and the peak transmit power capability afforded by each active user's power budget.

2) Admission: The new users (circuit or packet) are selected and allocated transmit power levels to optimize some measures such as the throughput.

The Lowest Power Algorithm (LPA) Ayyagari et al. [35] determines the ratio of peak received power to minimum received power for each mobile user and then selects the link with the smallest ratio to scale all the minimum link powers. The obtained results are compared with the Maximum Power Algorithm (MPA) where all active users operate at their maximum power. It is shown that the interference margin and the data rate available for a new code is higher for LPA.

In Chen et al. [36], transmission power control and channel admissions are managed jointly in order to maintain the SIR's of all links above the required quality factor at all times. Two algorithms that guarantee the quality of connection for active users and predict the effect of a possible new mobile on the SIR of active mobiles by measuring the path loss among the stations are proposed. In the second approach, instead of a global controller that collects and process information for call admission and power assignment, a local control decision is used. The problem is formulated as to find a basic feasible power vector solution to activate a mobile. This controls the activation of a mobile and the assignment of transmission power to it.

In Huang et al. [37], two schemes are proposed: a transmitted power based call admission control (TPCAC) scheme that protects ongoing calls and a received power based call admission control (RPCAC) scheme that blocks new calls when the total power received at a base station exceeds a threshold. A weighted combination of call blocking and call dropping probabilities is used as a system performance measure. The simulation results show that the total received power provides a more useful measure of system load than user's transmitter powers and that RPCAC algorithm provides significant reductions in the weighted combination of call blocking and call dropping.

In Akyildiz et al. [38], a novel medium access control protocol called wireless multimedia access control protocol with BER scheduling for slotted CDMA based systems is proposed. The idea is that if multimedia data is transmitted with the conventional power control schemes where the capacity is limited by the traffic with the lowest BER requirements, it will be wasteful to schedule simultaneous transmissions of voice and data packets with BER requirements on the order of 10^{-3} and 10^{-9} respectively. It schedules the transmission of multimedia packets according to their BER requirements so that packets with equal or similar BER requirements are transmitted in the same slots. The main objective is to maximize the throughput and to minimize the packet losses. Since packets are classified according to their BER requirements, the conventional power control scheme can be used with one power level for each slot which is simple to implement.

In Sun et al. [13], two call admission schemes are implemented; they are based on the outage probability. In predictive call admission control, a linear predictor is used to predict the future traffic from its present and past values. If the call setup time is smaller regarding to the traffic variations, advantage of predictive algorithm is negligible.

In Wu et al. [39], the measured traffic is used to control transmission power distribution in a wireless CDMA system for Poisson traffic model. It is assumed that the required bit error rate for media with high transmission rates (higher power and lower processing gain) needs to be lower than the media with low transmission rates (lower power and higher processing gain). Two types of service are considered as media with high priority (e.g., telephone and video) and media with low priority (e.g., data and fax). An optimal power distribution is computed to provide the required BER of media with high priority and achieve the maximum throughput and minimum BER of media with low priority.

A hybrid channel assignment scheme is proposed in Kim et al. [20] for slotted DS-CDMA systems for integrated voice/data services where voice traffic is transmitted in circuit mode while data traffic is transmitted in packet mode. It is assumed that data packet length is equal to one subslot length. Optimum power levels are found for each time slot by considering the number of voice calls and the number of data bursts.

In Fantacci et al. [21], the performance evaluation of voice and data services in wideband slotted DS-CDMA systems is derived for optimum code allocation at the base station. The analysis is done under the assumption of a Poisson message arrival process and geometric message length distribution and that mean message delay is equal to mean packet slot delay.

In Yue et al. [23], the authors present the output and delay process analysis of slotted CDMA network system for integrated voice and data services where data users can use the voice codes during off-time of voice users. In their model, the time axis is slotted to segments corresponding to packet transmission time and it is assumed that at each slot maximum M (M=available number of codes) users can transmit successfully.

CHAPTER 3

SYSTEM MODEL

The system model consists of a CDMA cell with multimedia services. Two main classes are considered such as real time (class 1) and non-real time services (class 2):

Class 1: Delay intolerant application requiring a constant bit rate of R_i^1 bits/s and a bit error rate (BER) of at most Γ_i^1.

Class2: Delay tolerant users requiring a bit rate of at least $R_{i_{min}}^2$ bits/s and a bit error rate of at most Γ_i^2.

Each class is further divided into subclasses corresponding to different applications with given traffic characteristics and QoS parameters, e.g., both voice (class 1_1) and video (class 1_2) users correspond to the first class (Table 3.1). The traffic model of these classes, the communications channels of the CDMA system and the channel allocation schemes are explained in the following subsections.

3.1 Traffic Model

Users arrive to the system with Poisson arrival rate λ and the call/session durations are exponentially distributed with mean μ.

$$P(t) = \lambda e^{-\lambda t} \; t \geq 0, \lambda > 0 \tag{3.1}$$

$$P(s) = \mu e^{-\mu s} \; s \geq 0, \mu > 0 \tag{3.2}$$

Real time services

The accepted models for voice activity are the exponentially or geometrically distributed on-off models with mean on time $1/\kappa$ and mean off time $1/\beta$. The voice activity factor ξ can be calculated from the balance equations as $\xi = \frac{\beta}{\kappa+\beta}$.

	Class 1		Class 2	
Examples	Voice	Video	Web browsing	Email
Delay	bounded		sensitive	tolerable
Rate	guaranteed		not guaranteed	
BER	$\leq 10^{-3}$	$\leq 10^{-6}$	≈ 0	

Table 3.1 Traffic Classification

For exponentially distributed on-off times, the superposition of a constant number of N independent voice user can be modeled by a birth-date process as shown in Figure 3.1. The state probabilities are then:

$$P(j) = \binom{N}{j} \xi^j (1 - \xi)^{N-j} \tag{3.3}$$

If the number of active frames within "on" period and idle frames within "off" period are computed by a geometric random variable ($P(k) = p(1-p)^{k-1}$), the same process can be used when data rate is high relative to on-off times.

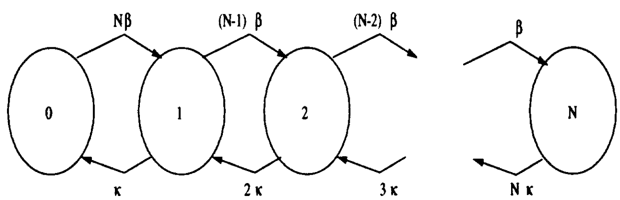


Figure 3.1 Voice traffic model

It is shown in Garrett et al. [40] that variable bit rate video transmission exhibits a self similar character and that the frame length conforms to a Pareto distribution , at least in the tail of distribution. The Pareto distribution has a probability mass function:

$$p(x) = \alpha\, k^{\alpha}\, x^{-\alpha-1}; \alpha, k > 0, x \geq k \tag{3.4}$$

If , $\alpha \leq 2$ then the distribution has infinite variance; if $\alpha \leq 1$ then the distribution has infinite mean.

Non real time services

Recently, the heavy-tailed packet length distribution has been widely used in many research and development studies for wireless technologies [41], Ajib et al. [69], Al Agha et al. [70], Yegenoglu et al. [71]. Exponentially distributed interarrival times and fixed, exponentially and Pareto distributed packet lengths with cutoff (maximum packet length) are implemented.

An exponential distribution with cutoff value m and mean μ_f is computed as:

$$f_X(x) = \begin{cases} \mu e^{-\mu x} & 0 < x < m \\ e^{-\mu m} & x = m \end{cases} \tag{3.5}$$

$$\mu_f = \frac{1 - e^{-\mu m}}{\mu} \tag{3.6}$$

A Pareto distribution with minimum (k) and maximum (m) values and shape parameter α is computed as:

$$f_X(x) = \begin{cases} \frac{\alpha k^{\alpha}}{x^{\alpha+1}} & k \leq x < m \\ (k/m)^{\alpha} & x = m \end{cases} \tag{3.7}$$

$$\mu_f = \frac{\alpha k - m(k/m)^{\alpha}}{\alpha - 1} \tag{3.8}$$

To simulate self-similar data such as Ethernet data, Pareto distribution is used for the interarrival times. Recent examinations of Ethernet, LAN traffic Leland et al. [45] and wide area network traffic Paxson et al. [46] have challenged the commonly assumed models for network traffic, e.g., the Poisson distribution. Measurements of real traffic indicate that significant burstiness is present on a wide range of time scales. It is shown that Pareto distribution describes well such types of traffics Nabe et al [47], Crovella et al. [48].

3.2 System Channels

Access channels: Access channels are channels used for communication of channel request and acknowledgment messages between the mobile and base station. The entire process of sending one request message and receiving (or failing to receive) an acknowledgment for that message is called an access attempt. One access attempt consists of one or more access sub-attempts delayed with a random delay and restricted by an impatience factor. In this study, the number of access channels is kept high so that an access failure occurs only when there is no available traffic channel for a new user (as a result of call admission). Transmission power through access channels and data rate are constant. The access request message (l_a bits) consists of class id and mobile id.

Control channels: Control channels are channels used for communication of power and rate control and acknowledgment messages between the mobile and base station. In the simulations, the number of control channels is high relative to number of users and transmission power through control channels and data rate are constant. Control message consists of cell id, mobile id and power and rate levels. ARQ (automatic request for retransmission) is applied for class 2 users where a transmission control sequence which request retransmissions of data received in error is implemented. ARQ message consists of cell id, mobile id, packet id and a flag indicating positive or negative acknowledgment.

Traffic channels: Traffic channels are used for data transmission between the mobile and base station. The number of traffic channels is determined by the number of transmission codes. Dedicated traffic channels carry information in a dedicated, point to point manner between a single mobile station and the base station. Common traffic channels carry information in a shared access, multipoint to point manner between multiple mobile stations and the base station. The frame size is l_{t_i} bits for i^{th} class. The attributes of the signal carried by the traffic channels are allocated power (P) and rate levels (R).

The allocation of these vectors is governed by the computation of the amount of interference in the traffic channels. Note that total noise can change within the packet segments; signal to noise ratio (SNR) is computed beginning from the first bit of the packet until the end of the reception of the last bit.

$$SNR(dB) = 10log\frac{P_r}{\eta_n} \tag{3.9}$$

$$\frac{E_b}{I_0}(dB) = SNR + 10logG \tag{3.10}$$

where G is the processing gain, P_r is the received power. The total noise (η_n) for the user in consideration is the sum of received power from other users and background noise (η_T):

$$\eta_T = \eta_b + \eta_a \tag{3.11}$$

$$\eta_b = (T_r + T_b)WB \tag{3.12}$$

where T_r and T_b are receiver and background temperatures respectively, B is the Boltzmann constant, W is system bandwidth and η_a is the ambient noise level.

Received power at cell site is computed as:

$$P_r = P_{in}g_t p_l g_r \tag{3.13}$$

where P_{in} is the inband transmitted power, p_l is the path loss and g_t and g_r are the transmitter and receiver antenna gain respectively.

$$
\begin{aligned}
P_{in} &= P_t(b_{max} - b_{min})/W && (3.14)\\
b_{max} &= min(f_t + W, f_r + W) && (3.15)\\
b_{min} &= max(f_t, f_r) && (3.16)
\end{aligned}
$$

where P_t is the transmitted power and f_t and f_r are the transmitter and receiver base frequencies respectively. Path loss is computed as:

$$
\begin{aligned}
p_l &= (\frac{\lambda}{4\pi d})^{\gamma} && (3.17)\\
\lambda &= \frac{c}{t_f + W/2} && (3.18)
\end{aligned}
$$

where c is the light velocity, d is the distance between the user and the cell site and γ is the propagation constant. The computed error rate for the packet is then compared with given Γ value to decide the acceptance of the packet receipt.

This study focuses on the optimal allocation of traffic channels with the corresponding power and rate vectors that meet the user QoS requirements. According to the allocation control at session, burst and packet level, the transmission modes are distinguished as dedicated circuit, burst/packet transmission and common channel packet switching. In the next sections, these techniques are investigated for different media services of the same class.

3.3 Switching Methods

In this section, different switching techniques are described. Figure 3.2 provides the state diagram of the circuit and dedicated burst/packet transmission modes.

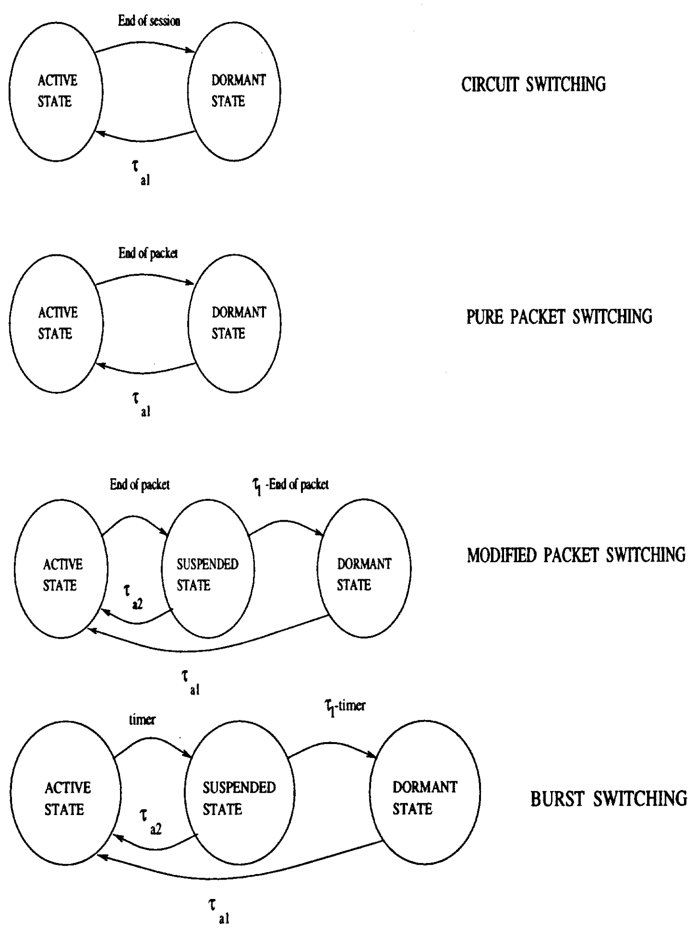


Figure 3.2 State diagram of switching techniques

3.3.1 Circuit Switching

In circuit switching, the users are allocated a dedicated channel and a continuous connection is guaranteed during a session. Each session arrival is either allocated to a dedicated channel or blocked after an access delay (Figure 3.2). A blocked user is lost if the waiting time exceeds the tolerance time. The blocking probability and waiting times can be calculated for different system parameters as defined in Barrer et al. [49].

The user first sends a request message containing the information of user's class through an access channel. The cell site decides to accept or reject the user according to the active users QoS requirements. If the user is accepted, an acknowledgment message with the assigned pn code and the required power level and data rate is send to the mobile and the active user list is updated. Each cell is assigned a long pn code and each admitted user is assigned a short code within the cell. If a negative acknowledgment is received or no response is received within a predetermined time interval, the user resends the request message after a random delay. If the allowed number of reaccess attempts is done and no positive acknowledgment is received, the user is blocked and cleared from the system.

Upon receiving the channel request message, the base station computes the new power level for each active user and the new user according to its required Γ value. If feasible power and rate vectors are found, the new user is accepted to the system. Then, the power and rate adjustment messages are sent to the active users through the control channels. A power vector is feasible if for every active link, a positive transmission power is found when there is no peak transmit power constraint or a transmission power level smaller than the peak transmit power when there is a peak transmit power constraint for the mobile. Similarly, a rate vector is feasible if for every active link a rate level greater than the minimum required rate can be assigned ($0 \leq P_i \leq P_i^{max}$ and $R_i^{min} \leq R_i \; \forall i = 1...N$).

If the user receives a positive acknowledgment from the cell site, it starts to transmit the packets through the traffic channels with the assigned code and required power and rate level. When circuit switching is applied, the user transmits until the end of the call when the user sends a call termination packet to the cell site. The base station then updates the active user list and the statistics for QoS evaluation.

Minimization of Total Transmitted Power For Circuit Switching Method

This section displays the circuit switching results for a single cell when the power and rate allocation is done according to the analysis given in Section 2.1.

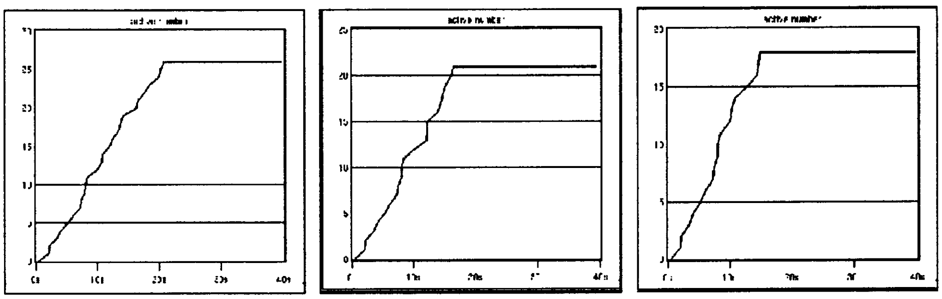


Figure 3.3 Results of call admission by minimizing total transmit power for two subclasses of real time service; $W = 1.25$ MHz, $R_1 = R_2 = 9600$ bps, $\Gamma_1 = 5.019, \Gamma_2 = 10, \gamma = 4$,: Left: All voice users ; Middle: $N_1 = 15$; Right: $N_1 = 10$.

Figure 3.3 displays simulation results where N_2 class 1_2 (video) users are accepted when N_1 class 1_1 (voice) users are already present. The horizontal axis is the observation time and vertical axis is the number of users accepted to the system ($N_1 + N_2$). Using the formulas given in section 1 maximum number of class 1_2 users that can be accepted is computed as:

$$N_2 = \lfloor 1 + W/(R_2\,\Gamma_2) - N_1\,\frac{(W/(R_2\,\Gamma_2)+1)}{(W/(R_1\,\Gamma_1)+1)}\rfloor \tag{3.19}$$

Figure 3.4 displays possible admission of voice and video users for the parameter set given in Figure 3.3.

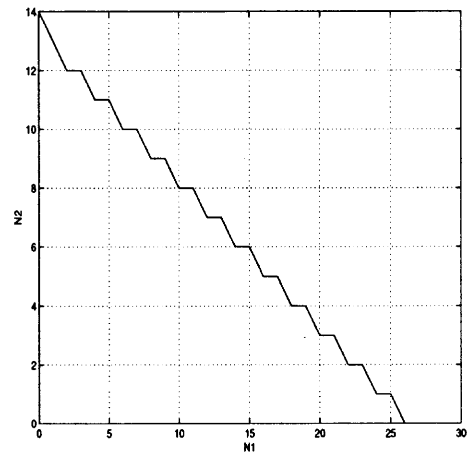


Figure 3.4 Admission intervals

3.3.2 Packet Switching

In modified packet switching, a user accesses to the system from idle state at the beginning of the session (Figure 3.2). In the suspended state, the user does not transmit but the state information is maintained by both the base station and the user [42]. The user in the suspended state access to the system after an access delay τ_{a2}. Once in the suspended state, if the inactivity period extends beyond a certain limit (t_1) the user makes a transition to the inactive idle state.

3.3.3 Burst Switching

Recent traffic analyses from various networks show the inefficiency of conventional channel allocation schemes for the integration of multimedia capabilities into mobile communications. The collected and analyzed traffic measurements in Crovella et al. [48], [50], Leland et al. [45], Paxson et al. [46] provide two results that are crucial in channel allocation: i) channel allocation schemes that are optimal for traditional Markovian models are not appropriate for multimedia packet networks, ii) long-range dependence in bursty traffic with fractal dimensions is a dominant factor in network system behavior e.g., packet loss and delay characteristics.

Circuit and packet switching techniques are insufficient in meeting the quality of service (QoS) requirements of bursty long multimedia messages due to the poor channel utilization and high per packet delay respectively. The proposed burst switching technique in cdma2000 MAC layer ([42]) aims to overcome these problems by allocating the dedicated channels to the burst of data and releasing them at the end of the bursts. This ideal burst switching system would immediately release the circuit at the beginning of the idle period following the packet burst, so that the allocation delay constraints would be satisfied, while channel utilization is maximized Ozer et al. [52]. In Taaghol et al. [53],[54], an hybrid multiple access protocol based on burst reservation technique is studied for different systems.

If the user is accepted to the system, a dedicated channel is assigned to that user. The mobile user can be in three different states: “active state” where it transmits data through a dedicated traffic channel, “suspended state” where only control channels are hold and “dormant state” where the user is assumed to be idle. The state transitions are controlled by the base station via “timer” values. The base station starts a timer for each active user and resets it when there is a new packet arrival from the corresponding user. If the inactivity period extends beyond a certain timer the user enters to the suspended state and then either makes a transition to the inactive idle state if this inactivity period extends beyond another timer value or restarts to transmit. Base station keeps two lists for users in active state and for users in suspended state. The user from suspended state tries to reaccess the system by sending an access message that contains only the mobile id. Optimal timer that determines the burst length is dependent on user traffic characteristics, time scale of interest, system load and QoS requirements. Note that in burst switching case, call admission must not only be dependent on active users but also to suspended state users that can reaccess to the system (in order to foresee the potential to admit a new user).

3.3.4 Common Channel Packet Switching

Common Packet Channel (CPCH) mechanism has been shown to be an efficient transfer mechanism of packet data in wireless environments for non-real time applications such as E-mail, HTTP and FTP [68]. CPCH message transmission portion operates in power controlled CDMA. Each message can have variable length where the maximum length is a higher layer parameter [67]. Since error control via acknowledgments and retransmission in non-real time applications is crucial, especially in the environments where message losses are usually higher, a retransmission scheme is used. The additional delays caused by retransmissions are likely to be tolerable in applications with less stringent delay bounds, while the loss of some of the messages is often intolerable since completeness of information delivery is essential.

Users access to the system by sending a message that contains the information of user's class as in circuit switching. However, there is no transmission control depending on interference level during the call. The packet transmission takes place on a common fixed-rate channel with different codes. A positive or negative acknowledgment is sent to the user after the reception of each packet by comparing the packet error with the predetermined threshold value. If negative acknowledgment is received or no response is received within a predetermined timeout value, the user retransmits the packet after a random delay. All other packets wait in the mobile buffer until previous packet is transmitted successfully.

CHAPTER 4

COMPARISON OF SWITCHING TECHNIQUES

In this chapter, different switching techniques explained in the previous section are compared via analysis and simulation for different traffic characteristics. First, circuit mode and burst/packet mode transmission techniques are studied. The following two sections discuss the advantages of burst switching technique for bursty data traffic for unslotted and slotted CDMA networks respectively. Then, the comparison of burst mode transmission and common channel transmission schemes is done for integrated services for different data packet length distributions.

4.1 Unslotted CDMA Wireless Networks

4.1.1 Traffic Model

Within a session, packet interarrivals are independent, identically distributed according to a Pareto distribution with packet arrival rate $\lambda_p = (\alpha - 1)/(k\,\alpha)$ (Section 3.1). Hence, the packet arrival process has strong burstiness due to the heavy tailed Pareto distribution. The packet sizes are 53 bytes. The sessions arrive with a Poisson arrival rate λ_u and have exponentially distributed duration of mean $1/\mu$. The pool of users is assumed to be infinite.

4.1.2 System Model

CDMA is used as an unslotted multiple access technique where N simultaneous transmissions are assumed to be possible. It is assumed that users have the same type of traffic load and collisions occur only when more than N users attempt to transmit (N-channel model Yin et al. [19]). Two types of delay are considered: access delay and transmission delay. Access delays vary according to the switching techniques and are described below. The transmission delay per packet is equal to the number of chips per packet divided by the transmission time of a chip (1.2288 Mc/sec) where the number of chips per packet is packet size multiplied by the number of available CDMA codes.

In the following the notation used throughout this section is given:

- λ_u: Arrival rate of users.
- $1/\mu$: Mean of session duration.
- ρ:$\lambda_u/(N\mu)$=traffic load of sessions
- λ_p: Arrival rate of packets within a session.
- α: Pareto shape parameter.
- k: Pareto parameter for minimum interarrival time.
- N: Number of simultaneous transmissions without collision
- τ_{a1}: Access delay for a packet from "idle state"
- τ_{a2}: Access delay for a packet from "suspended state"
- τ_1: Packet interarrival time threshold for state transition
- τ_t: Transmission delay for a packet
- *timer*: Timer for burst switching

In order to show the improvement achieved by burst switching in resource utilization and QoS of users, the switching techniques are compared for the bursty traffic data. To recall the definitions of the transmission modes, in circuit switching, the users are allocated a dedicated channel and a continuous connection is guaranteed during a session. Each session arrival is either allocated to a dedicated channel or blocked after an access delay (Figure 3.2).

In pure packet switching, the user starts to transmit after an access delay τ_{a1} (Figure 3.2). The user transmits successfully if there are less than N packets being transmitted at the request time, otherwise $N+1$ users are backlogged with a uniform random delay [0,1]. If a new packet of a backlogged user arrives, this packet enters to the queue.

In modified packet switching, a user accesses to the system from idle state at the beginning of the session (Figure 3.2). It is assumed that the user starts to transmit an access delay τ_{a1}. In the suspended state, the user does not transmit but the state information is maintained by both the base station and the user [42]. The user in the suspended state access to the system after an access delay τ_{a2}. Once in the suspended state, if the inactivity period extends beyond a certain limit (t_1) the user makes a transition to the inactive idle state.

In burst switching, the dedicated channel is deallocated upon the transmission of the packet burst i.e., if the inactivity period extends beyond a certain "timer". At that time, the user enters to the suspended state and then either makes a transition to the inactive idle state if this inactivity period extends beyond t_1 or restarts to transmit. The dedicated channel is allocated if there are less than N users that are holding channels at the request time, otherwise $(N+1)^{th}$ users packet is backlogged with a uniform random delay [0,1]. If a new packet of a backlogged user arrives, this packet enters to the queue.

4.1.3 Burst Estimation

In this section, an optimal "timer" that detects the beginning of an idle time period in a session is investigated. The aim is to maximize the utilization while meeting the channel allocation delay constraints. As a result of the assumptions described in the simulation model, $\tau_{a1} + \tau_t < k < timer < \tau_1$. Then, the mean value of access delay of a packet except the first one is:

$$\left(\frac{k}{\tau_1}\right)^{\alpha} \tau_{a1} + \left(-\left(\frac{k}{\tau_1}\right)^{\alpha} + \left(\frac{k}{timer}\right)^{\alpha}\right) \tau_{a2} \tag{4.1}$$

The mean value of idle time within the user's channel holding is:

$$\left(\frac{k}{timer}\right)^{\alpha} timer + \frac{\alpha\ k^{\alpha}\ timer^{1-\alpha} - \alpha\ k}{1-\alpha} \tag{4.2}$$

Hence, the utilization i.e. , the ratio of packet duration to the channel holding time is:

$$\frac{\tau_t}{\tau_t + \left(\frac{k}{timer}\right)^{\alpha} timer + \frac{\alpha\ k^{\alpha}\ timer^{1-\alpha} - \alpha\ k}{1-\alpha}} \tag{4.3}$$

Similarly, the following values are computed to compare the performance of the timer:

The mean value of time interval between the channel de-allocation and re-allocation is:

$$\frac{k^{\alpha}\ timer^{1-\alpha}}{\alpha - 1} \tag{4.4}$$

The mean value of number of packets in a burst is:

$$\left(1 - \left(\frac{k}{timer}\right)^{\alpha}\right) \left(\frac{k}{timer}\right)^{-\alpha} \tag{4.5}$$

The variance of number of packets in a burst is:

$$\left(1 - \left(\frac{k}{timer}\right)^{\alpha}\right) \left(\frac{k}{timer}\right)^{-2\alpha} \tag{4.6}$$

Utilization and average delay versus timer in burst switching are displayed in Figure 4.1 and in Figure 4.2 for different packet arrival rates and for different Pareto shape parameters respectively. Note that the same results are obtained via simulation with two characteristics: slow convergence to steady state and high variability at steady state Crovella et al. [55].

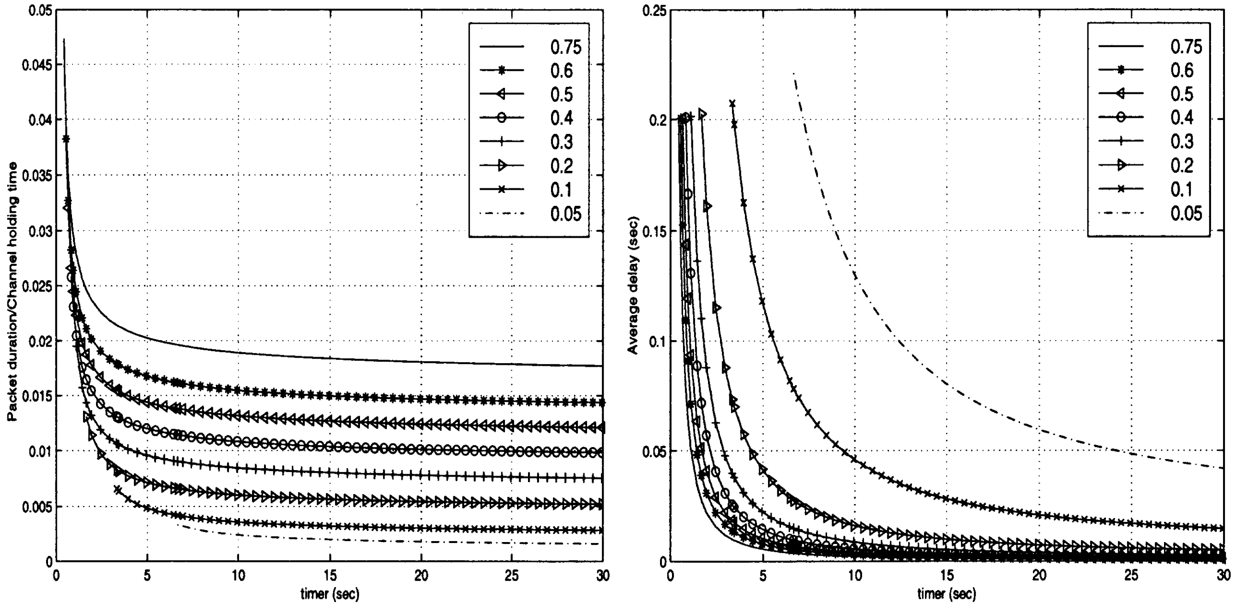


Figure 4.1 Utilization and average delay vs timer for different packet arrival rates λ_p=0.05 $\sim$ 0.75 packet/sec. (Pareto shape parameter = 1.5). Left: Ratio of packet duration to channel holding time vs timer ; Right: Average access delay vs timer.

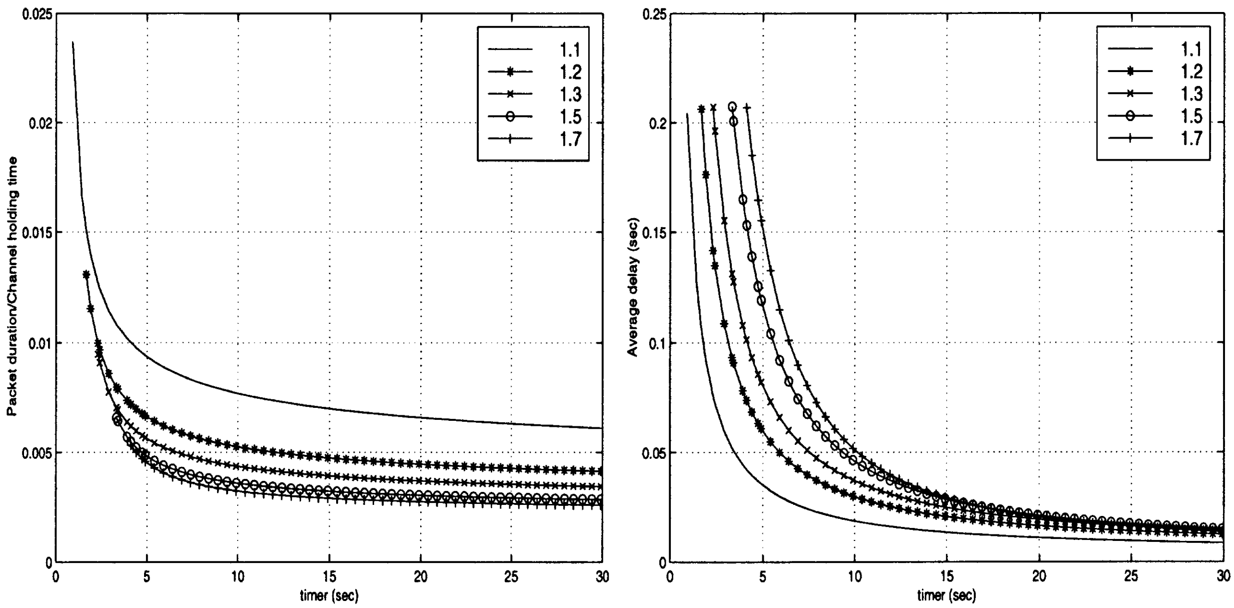


Figure 4.2 Utilization and average delay vs timer for different Pareto shape parameters α = 1.1 $\sim$ 1.7 (Packet arrival rate = 0.1 packet/sec). Left: Ratio of packet duration to channel holding time vs timer ; Right: Average access delay vs timer.

Figure 4.3 shows the utilization and mean access delay versus timer values for $\alpha = 1.5$ and $\lambda_p = 0.5$ packets/sec/user. The utilization maximizes when timer is higher while mean access delay minimizes when timer is smaller. Since timer can not be smaller than k, the maximum utilization is achieved when timer is equal to k. It can be seen that after some value, increasing timer do not change significantly the utilization and mean access delay. The optimal timer value can be chosen to minimize the total penalty given to the utilization and access delay or to maximize the utilization while satisfying a given delay constraint. Also, the performance of burst switching can be evaluated according to some criteria:, e.g., timer should maximize time interval between the channel de-allocation and re-allocation, i.e., should not divide packet bursts.

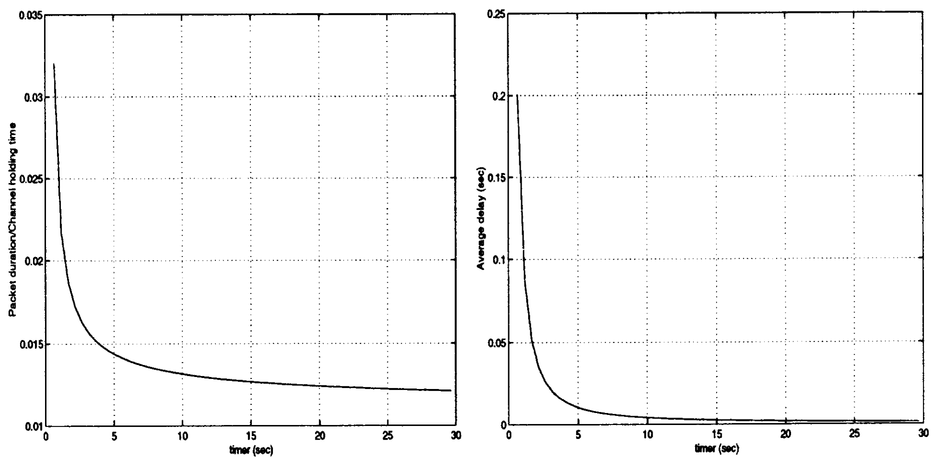


Figure 4.3 Utilization and mean access delay versus timer values for $\alpha = 1.5$ and $\lambda_p = 0.5$ packets/sec

If the constraint is that 95% of packets within a session have no access delay, then the timer value that satisfies this constraint is $0.05^{-1/\alpha}k = k\,0.05^{k\lambda_p - 1}$ that depends on the packet arrival rate and minimum packet interarrival time. For the example given in Figure 4.3, the channel allocation and de-allocation times are

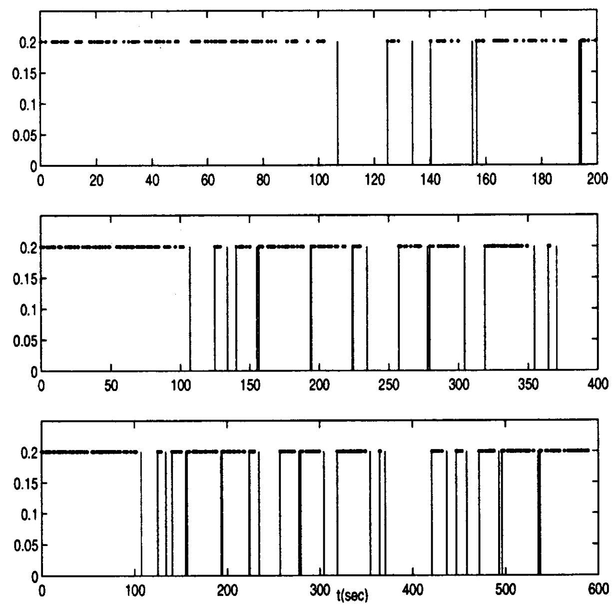


Figure 4.4 The channel allocation and de-allocation times are shown with lines for $\alpha = 1.5$,$\lambda_p = 0.5$ packets/sec, *timer* = 4.9120 sec

shown in different time scales for the timer value that provides the delay constraint (Figure 4.4).

The next set of experiments is performed to show that optimal timer value found for one session is also optimal for a system of multiple users with the same type of traffic when the probability of collision is not high enough to effect the percentile delay constraint. This timer satisfies the percentile delay constraint while maximizing the channel utilization.

For the following parameter set, the circuit, packet and burst switching techniques are implemented for infinite number of users: α: 1.5, τ_{a1}: 0.4 sec., τ_{a2}: 0.2 sec., τ_1: 15 sec.

Figure 4.5 shows the probability of session loss and waiting time per session for different traffic loads and utilization per channel, i.e., the ratio of packet duration to the channel holding time for different packet arrival rates. Note that, when λ_p

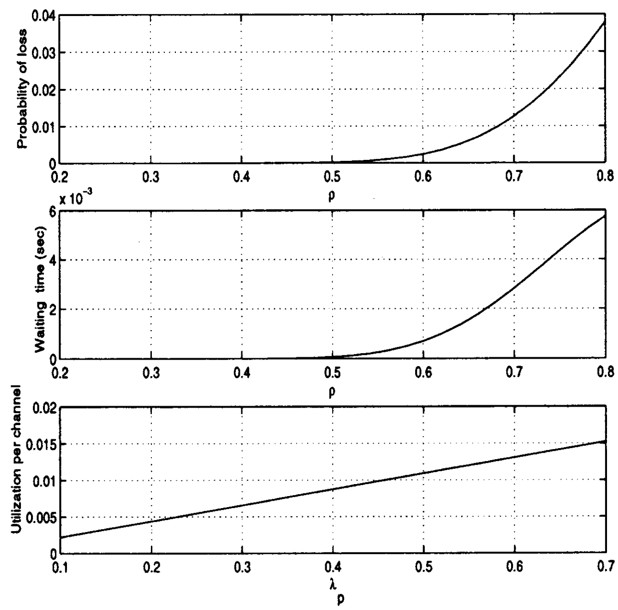


Figure 4.5 Circuit switching results for different traffic loads

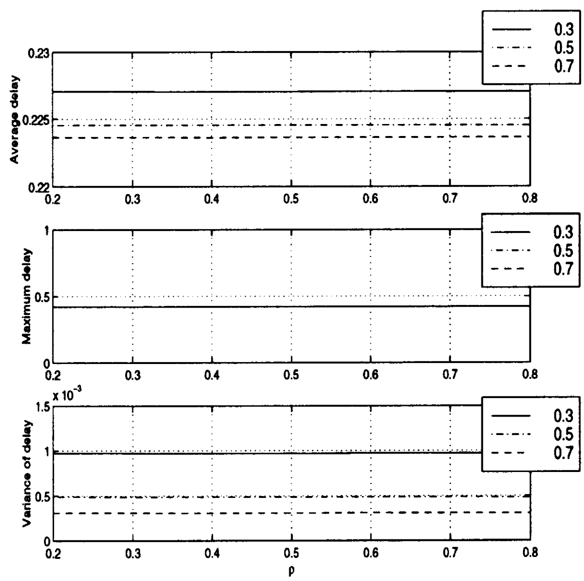


Figure 4.6 Packet switching results of $\lambda_p = 0.3, 0.5, 0.7$ packets/sec/user for different ρ; the delay values are in seconds.

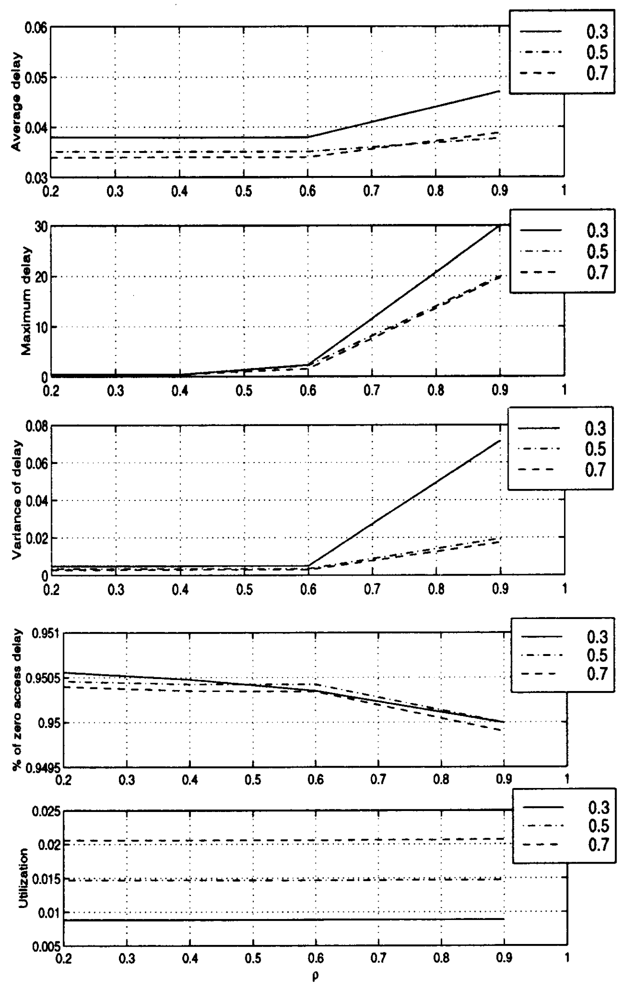


Figure 4.7 Burst switching results of $\lambda_p = 0.3, 0.5, 0.7$ packets/sec/user for different ρ; the delay values are in seconds.

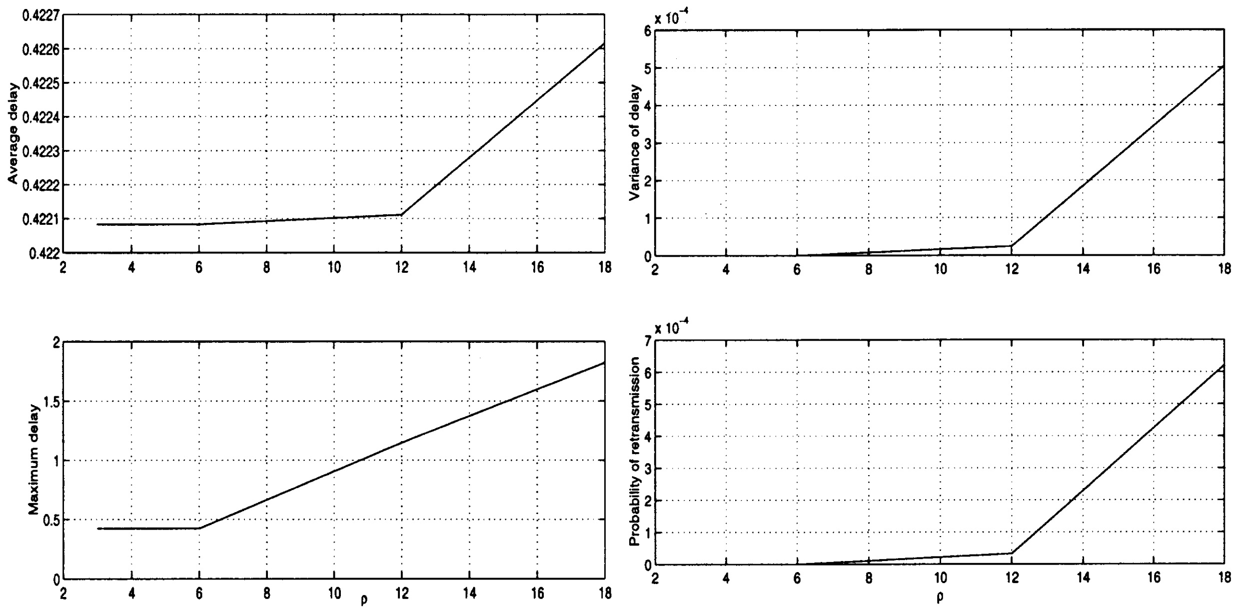


Figure 4.8 Pure packet switching results for different ρ; λ_p=0.5 packet/sec/user.

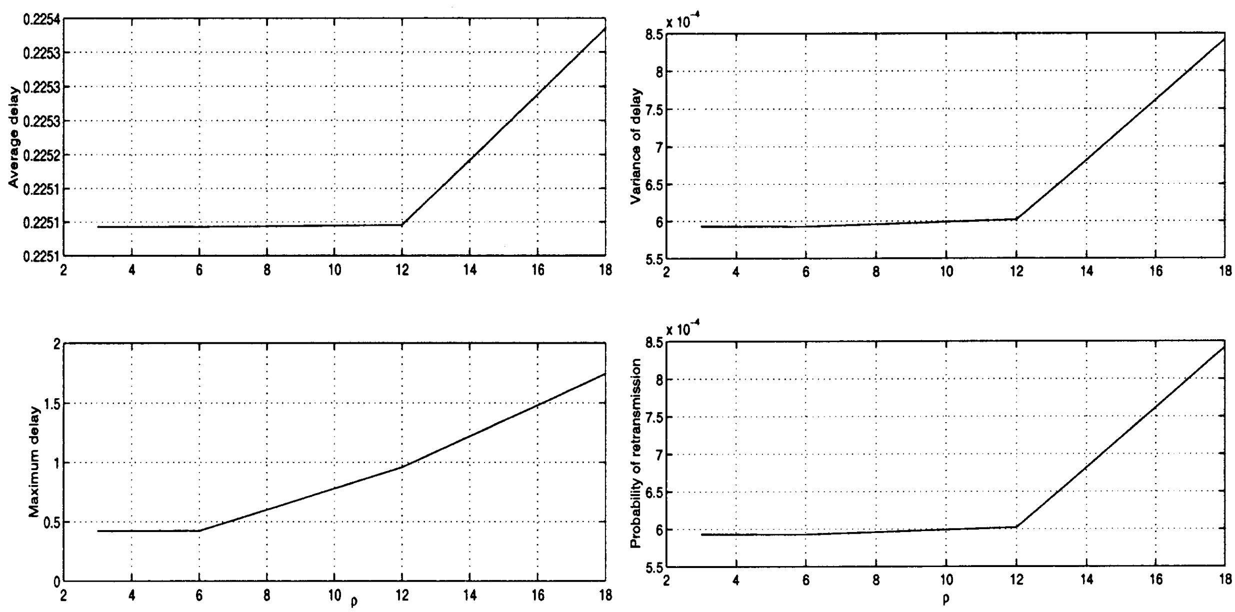


Figure 4.9 Modified packet switching results for different ρ; λ_p=0.5 packet/sec/user.

increases for a given session duration the utilization increases since more packets are generated reducing idle part of channel holding time.

Figure 4.6 shows the delay (access and transmission delay) characteristics of a packet for different traffic loads for modified packet switching technique. Since the system shows the same characteristics as the system with one user for the given traffic load, we can see that the simulation results for average delay are close to the analytic results: Average delay of a packet except the first one is $(\frac{k}{\tau_1})^\alpha\ \tau_{a1} + (1 - \frac{k}{\tau_1})^\alpha\ \tau_{a2}$, that are 0.226 sec, 0.224 sec and 0.223 sec for $\lambda_p = 0.3, 0.5, 0.7$ packets/sec/user respectively. The average delay in pure packet switching for the same traffic load is $\tau_{a1}+\tau_t$=0.422 sec. The utilization per channel in both techniques , i.e., the ratio of packet duration to the channel holding time is 1 and minimum access delay per packet is 0.2 sec and 0.4 sec for modified and pure packet switching respectively.

Figure 4.7 shows the delay (access and transmission delay) characteristics of a packet and utilization for different traffic loads for burst switching technique. Timer values for λ_p=0.3,0.5,0.7 packet/sec/user are 8.1867 sec, 4.9120 sec and 3.5086 sec respectively. The results of simulations show that the optimal timer found for one user satisfies the delay constraint while providing an efficient channel utilization for a system of multiple users when blocking probability is small. The improvement on channel utilization can be seen by comparing the results with the results of session loss and utilization in Figure 4.5. However, when ρ is large, two characteristics of the heavy tailed distribution mentioned above are observed; slow convergence to steady state and high variability at steady state in delay analysis.

The following set of experiments is performed to show the effect of session traffic load on the techniques. Note that the circuit switching is not stable in this case and packet switching techniques can not satisfy the access delay constraint (Figure 4.8,4.9). In this case the total delay in packet switching is increased in average with probability of retransmission times average backlogging delay. Figure 4.10 shows the delay characteristics of burst switching under heavy traffic for $\lambda_p = 0.5$ packets/sec/user and $\rho = 1.6667$. The optimal timer was previously found as 4.912 sec for 95% zero access delay constraint. In this example, high variability in the delay characteristics is observed (Figure 4.10). In contrast to the short tailed distributions, the probability of very large deviations is non-negligible. However, it can be seen that the percentage of packets that have high variable delay characteristics do not change significantly the percentile delay constraint in average.

This section introduces the ideal burst switching technique as an optimal channel allocation scheme for third generation cdma systems. It is shown that burst estimation in a user session can be achieved by optimizing a timer value to release the channel when idle time period begins. This timer depends on the packet arrival rate and minimum interarrival time of a packet. This timer value maximizes channel utilization under an access delay constraint.

4.2 Slotted CDMA Wireless Networks With Aggregated Traffic

In this section, the performance simulation results and analytical bounds on packet delay and channel utilization for burst switching in slotted CDMA are given.

4.2.1 System and Traffic Model

Consider a slotted CDMA based wireless packet data system where C users generate the same type of traffic and request the same type of quality of service (QoS) can be served simultaneously. The system is investigated for a time interval corresponding to the session duration $(1/\mu)$ of N $(0 < N < \infty)$ simultaneous user transmissions. The total arrival process to the system is assumed to be stationary where $A(s,t)$ denotes the amount of packet traffic arriving in the time interval $(s,t]$. Packet interarrival times for each user within a session are assumed to be independent , identically distributed according to Pareto distribution with shape parameter α and location parameter k as the minimum interarrival time (Section 3.1). α is chosen to be between 1 and 2. Hence, the packet arrival process has strong burstiness due to the heavy tail. In Gordon et al. [56], it was shown that the packet count of arrival processes with heavy-tailed interarrival time distribution, e.g., Pareto distribution is asymptotically self similar.

The aggregate channel is formed by a superposition of N users' session. As stated in Taqqu et al. [57], using Mandelbrot's construction of FBM, a self similar process may be generated by superimposing many simple renewal reward processes, in which the rewards are restricted to the values of 0 and 1, and in which the inter-renewal times are heavy tailed. In this study, 1/0 values indicate the active/inactive period of the user for the corresponding time unit. As the number of such sources (N) grows large, the construction yields a self-similar fractional Gaussian noise process with Hurst parameter $H = (3 - \alpha)/2$. In Leland et al. [45], it was suggested that a simple renewal reward process is an adequate traffic source model for an individual Ethernet user and the aggregation results in fractional Brownian motion (FBM) when packet interarrivals are heavy tailed. In Paxson et al. [46], a similar approach is used where they show that the resulting process is pseudo self-similar.

The self-similar arrival process $A(s,t)$ is determined by three parameters: a mean rate $m > 0$ that measures the quantity of traffic, a Hurst parameter $H \in [0,1]$ as the rate for decay of correlations, and a variance parameter σ^2 or peakedness parameter $a = \sigma^2/m$ to describe the magnitude of fluctuations around mean Neidhardt et al. [58]. The last two parameters refer to the burstiness of the traffic Beran et al. [59] as $E[A(s,t)] = m(t-s)$, $Var[A(s,t)] = V(t-s) = \sigma^2|t-s|^{2H}$.

The increment process $Z = (Z(k) = A(k, k+1) : k \geq 0)$ is called fractional Gaussian noise and is a stationary (discrete-time) Gaussian process with autocorrelation function $r(k) = 1/2(|k+1|^{2H} - 2|k|^{2H} + |k-1|^{2H}), k \geq 1$. Asymptotically, Z exhibits long range dependence and for $1/2 < H < 1$, $r(k) \sim H(2H-1)|k|^{2H-2}$. This process is characterized by the following properties: (i) $Z(k)$ has stationary increments; (ii) $Z(0) = 0$, and $E[Z(k)] = 0$ for all t; (iii) $E[Z(k)^2] = |k|^{2H}$ for all t; (iv) $Z(k)$ has continuous paths; (v) $Z(k)$ is Gaussian. From the packet traffic modeling viewpoint, the FBM traffic model is a reasonable representation of aggregate data traffic (i.e., formed by multiplexing a large number of independent data sources). The conditions under which the FBM model can be expected to be valid in practice are described in [50]; e.g., the traffic is aggregated from a large number of independent users, the range for time scales of interest coincides with the scaling region. In Norros et al. [60] it is shown that even the speed of convergence to a Gaussian process is rather slow theoretically, in reality, aggregate traffic of a limited number of sources can be modeled well by self-similar process, in particular as fractional Brownian traffic.

4.2.2 Burst Switching Technique and Performance

The time slots in a CDMA system correspond to the transmissions of ATM cells. In every time slot, C users requesting the same type of QoS can be served simultaneously. When the number of available channels (codes) C in the CDMA system is smaller than the number of simultaneous transmissions of sessions N, allocation of dedicated channels during the session (i.e., circuit switching on session basis) will increase the probability of loss for new users. Furthermore, this will lead to a poor channel utilization since the traffic has long idle times between the burst of data. This corresponds to a larger timer value while a smaller timer value imitates a switching system on a packet basis that will cause large allocation delays per packet. Note that burst switching is between these two extreme cases. A simple illustration of switching techniques for zero queue length is displayed in Figure 4.11 where the packet access delay and the timer value are taken as one slot time (t_s). The switching per session allocates dedicated channels to the users until the end of the session or until the interarrival of packets exceeds a predetermined threshold τ_1. A user is said to be in a dormant state in this case. Since the objective is to guarantee a continuous service for the whole session, τ_1 is chosen such that the probability of releasing the channel before the end of the session is kept small. Let the traffic model have the parameters $\alpha = 1.5$, $\lambda = 0.3$, $C = 30$ and $N = 80$, $\tau_1 = 180sec$, $1/\mu = 1800sec$. Then the probability that a user who grabs a channel will make a transition to the dormant state is $(k/\tau_1)^\alpha = 0.00048$. In this case, the loss probability of $N - C$ users that attempt to transmit after C users grabbed a channel, is close to 1. For the given parameters, this corresponds to a loss of 62.5% of users. Furthermore, the mean idle time between packets (except the minimum interarrival time) for these parameters is approximately $43sec$ and can be computed from

$$\tau_{idle} = \int_k^{\tau_1} \alpha k^\alpha x^{-\alpha-1} \lfloor x/t_s \rfloor t_s dx \simeq \frac{(\tau_1^{1-\alpha} \alpha k^\alpha - \alpha k)}{(1-\alpha)} \tag{4.7}$$

If the timer is smaller than or equal to the minimum interarrival time of the Pareto distribution then the switching is done for each packet. When the packet accesses the system, it is transmitted in the next corresponding slot if there are available channels otherwise it enters to the queue. The delay due to the synchronization of the packet arrivals to the slot time units is neglected. This system is analyzed by using the queuing analysis of FBM arrival processes. The analysis of queues driven by FBM arrival processes has mostly been developed in [50, 61] using Benes method. The virtual waiting time $V(t)$ in a queue with infinite waiting room, with service

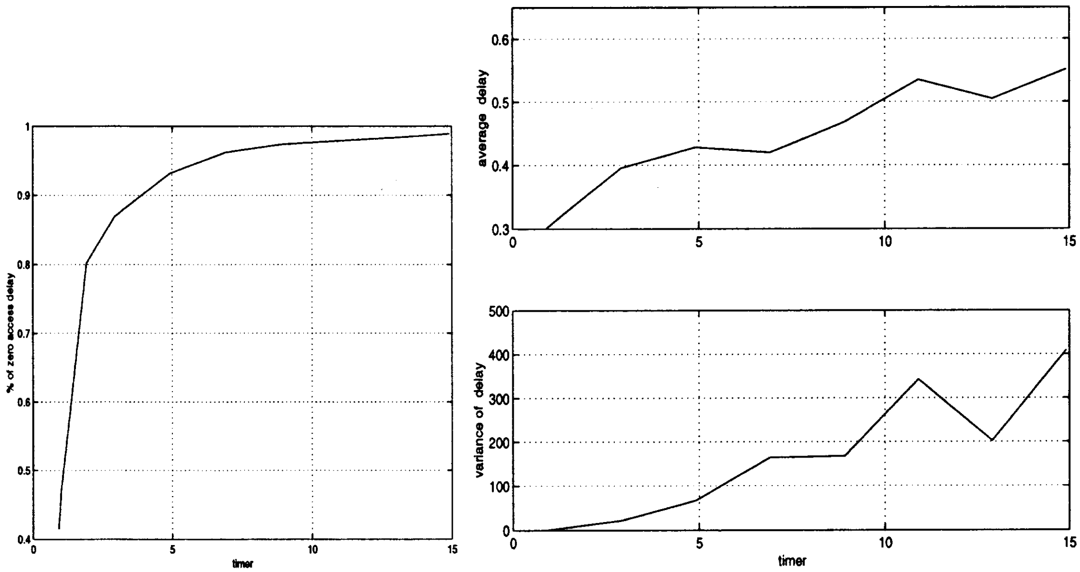


Figure 4.10 Burst switching results of λ_p = 0.5 packets/sec/user and ρ = 1.6667 for different timer values

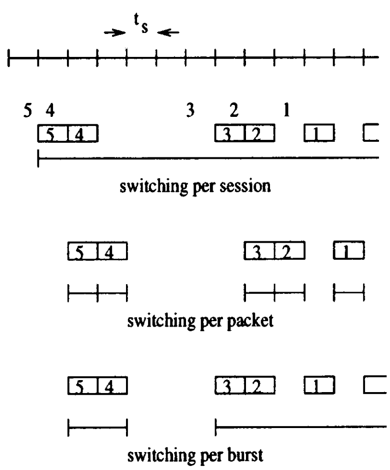


Figure 4.11 Illustration of switching techniques

rate equal to C and cumulative work arrival process $A(t)$ is calculated as

$$V(t) = \sup_{s \leq t}(A(t) - A(s) - C(t-s)) \tag{4.8}$$

where V(t) is non-negative and stationary and for $m < C$ it is almost surely finite. A lower bound on the tail of the complementary distribution can be obtained using the following inequality Norros [61]

$$\begin{aligned} P(V(t) > x) &= P(V(0) > x) \geq \max_{t \geq 0} P(A(t) > Ct + x) &\quad (4.9) \\ &= \max_{t \geq 0} \Phi\left(\frac{(C-m)t + x}{\sqrt{am}t^H}\right) &\quad (4.10) \end{aligned}$$

where $\Phi(.)$ is the complementary normal distribution. The maximum of $\Phi(.)$ can be obtained by setting the derivative of the argument of $\Phi(.)$ to zero

$$t = \frac{Hx}{(1-H)(C-m)} \tag{4.11}$$

Using this result and the approximation for the tail of the normal distribution result in the following lower bound

$$P(V > x) \geq \Phi\left(\frac{(C-m)^H x^{1-H}}{\sqrt{am}H^H(1-H)^{1-H}}\right) \tag{4.12}$$

$$P(V > x) \sim \frac{1}{\sqrt{2\pi}(1+\sqrt{cx})} exp(-cx^{2-2H}) \tag{4.13}$$

$$c = \frac{(C-m)^{2H}}{2am[(\frac{1-H}{H})^H + (\frac{H}{1-H})^{(1-H)}]^{-2}} \tag{4.14}$$

For $H > 0.5$, the asymptotic queue length distribution decays as a Weibullian in contrast to the exponential decay predicted by traditional, short range dependent traffic models. A qualitatively similar result was obtained in Parulekar et al. [62]. Figure 4.13 shows queue behavior of the system for switching on a packet basis. Note that the total delay per packet is the sum of access delay and the queuing delay. The access depends on the interarrival times of the user's packets to the system and the queuing delay can be computed using the above approximation. Since this is a FIFO system, it can be shown that the probability that a packet in the queue will be delayed more than q slots $P(W > q)$ is proportional to the probability of having more than qC packets in the queue (Figure 4.12). The mean access delay can be expressed as

$$\tau_{access} = (\frac{k}{\tau_1})^{\alpha}\tau_{a1} + (1 - (\frac{k}{\tau_1})^{\alpha})\tau_{a2} \tag{4.15}$$

where τ_{a1} and $\tau_{a2} < \tau_{a1}$ are the access delays if the user is in dormant or suspended state Dahlman et al. [63] respectively. The maximum total delay that a packet can have is then proportional to the probability $P(V > qC)$ and the access delay.

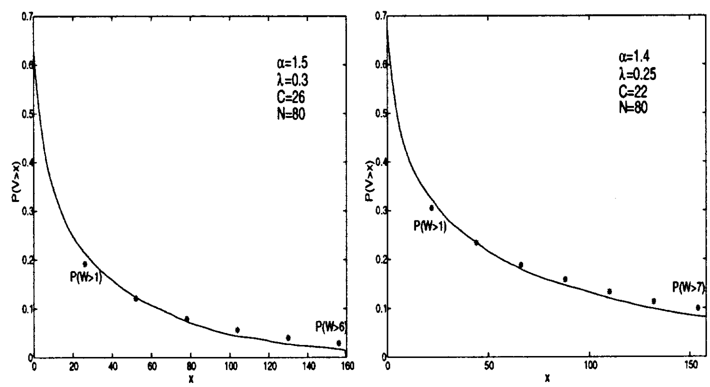


Figure 4.12 Experimental results of probabilities of queue length and waiting time for switching per packet.

The analyses for these two extreme cases of burst switching set the bounds for utilization of network resources and users' QoS characteristics as packet delay. The problem is then to find an optimal timer τ_2 that meets the user QoS requirements while increasing channel utilization. Note that there is a tradeoff in the choice of timer value, as larger timer values increase the user loss probability (due to unacceptable waiting delays) and this introduces a poor channel utilization while smaller timer values increase the packet access delay.

The timer value that meets the QoS requirements as packet delay and that maximizes the throughput will decrease the mean idle time

$$\tau_{idle} \simeq (\frac{k}{\tau_2})^{\alpha}\tau_2 + \frac{(\tau_2^{1-\alpha}\alpha k^{\alpha} - \alpha k)}{(1-\alpha)} \tag{4.16}$$

The access delay is then

$$\tau_{access} = (\frac{k}{\tau_1})^{\alpha}\tau_{a1} + (-(\frac{k}{\tau_1})^{\alpha} + (\frac{k}{\tau_2})^{\alpha})\tau_{a2} \tag{4.17}$$

If the QoS is defined in terms of mean packet delay, then the optimal timer can be chosen by minimizing the average delay. This optimization depends on the parameters of the arrival process $A(t)$ for the corresponding time scale and the number of available channels (C). Note that the queue length (virtual waiting time) increases with the timer value while the access delay decreases. Hence, the optimal timer value must satisfy the overall packet delay constraint. The simulation results of this minimization are displayed in Figure 4.14 where the system is observed for $1/\mu = 1800sec$. If the QoS is defined as a percentage of packets that can have a predefined maximum delay, then the timer will depend on the probability of queue length rather then the mean values. The queue behavior of the system for different timer values are given in Figure 4.15. For larger timer values, the probability that a packet will find the system busy increases, however the gain obtained from the access delay will also increase. Given the parameters of the arrival process, the required number of channels for meeting the QoS demands of the users can be driven using the experimental results. It can be seen that for the same arrival mean, the tail of the queue length is heavier when the interarrival time distribution has heavier tail (when α is smaller). Note that, the queue behavior of burst switching is different from the bounds obtained using Benes method $P(V > x) \sim exp(-cx^{2-2H})$ since the service capacity C is no longer a fixed value. The number of available channels at any instant time unit (slot) depends on the interarrival time of the user packets since a channel is hold until the interarrival of a packet for that user exceeds the predefined timer value τ_2.

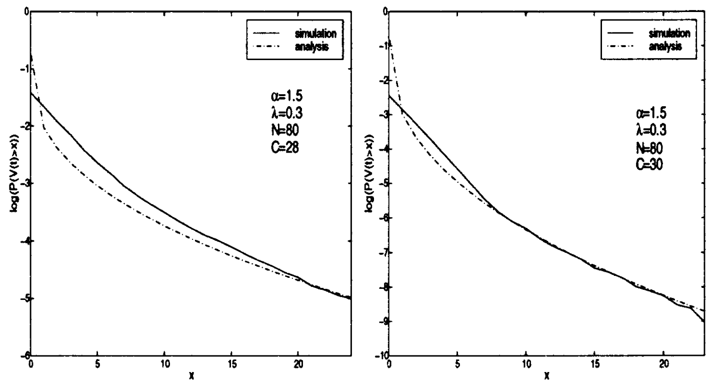


Figure 4.13 Queue length for switching per packet

In this section, burst switching technique is investigated for a slotted CDMA system. It is shown that burst switching decreases the packet access delay and increases the channel utilization. The bounds for extreme cases of burst switching (switching per session and switching per packet) in terms of packet delay (access and queuing delay) and idle time are derived. Experimental analysis of system characteristics for burst switching are given. It is shown that for a given arrival process and QoS requirements of the users, the system parameter (timer value) of the burst switching and the required number of channels can be obtained by minimizing the access and queuing delay per packet.

4.3 Comparison of Burst Mode and Common Channel Transmission Schemes for Integrated Services

4.3.1 Traffic Model

The system model consists of a CDMA cell with multimedia services. Two main classes are considered such as real time voice (class 1) and non-real time data services (class 2). The specific traffic characteristics and QoS requirements for each class are the models given in Chapter 3.

4.3.2 Simulation Model

A CDMA cell for uplink communication is simulated using Opnet network modeler where the communication between the base station and mobile terminals is done through access, control and traffic channels as explained in Section 3.2. The attributes of the signal carried by the traffic channels are: allocated power (P) and rate levels (R). The allocation of these vectors is governed by the computation of the amount of interference in the traffic channels. Note that total noise can change within a packet; signal to noise ratio (SNR) is computed beginning from the first bit of the packet until the end of the reception of the last bit by using the model given in Section 3.2 for the traffic channels. The computed error rate for the packet is then compared with given Γ value to decide the acceptance of the packet receipt.

4.3.3 Common Channel Packet Access Integrated with Voice Traffic

The voice users who are admitted to the system transmit their packets through dedicated traffic channels with fixed rate R_v. Their QoS requirement is to have BER smaller than 10^{-3} corresponding to a SNR value of 7dB for bpsk modulation. The data packets are transmitted through an unslotted spread spectrum packet radio network with fixed rate R_d. The data packet QoS requirement is assumed to be BER smaller than 10^{-6} corresponding to a SNR value of 10dB. If there is at least one erroneous bit, the packet is assumed to be lost. In this case, a negative acknowledgment is sent to the corresponding user through control channels. Backlogged user retransmits the packet after an exponential distributed transmission delay with mean $1/p$. The user can retransmit until a predefined maximum number of retransmissions. Packets are stored and served on a first-in-first out discipline. If the maximum buffer size is reached, the new coming packets are rejected. A copy of the packet in transmission occupies the top of the queue. The system model is illustrated in Fig. 4.16.

The system is simulated for the parameter set given in Table 4.1 for 10 voice and 10 data users. Fixed, exponential and Pareto (with different parameters, i.e. Pareto1 and Pareto2) data packet length distributions are used to show their effect on packet delay and loss characteristics. Pareto1 has parameters $k = 182, \alpha = 1.5, m = 12000$ for mean packet length 500 bits and $k = 278, \alpha = 1.5, m = 12000$ for mean packet length 750 bits. Pareto2 has parameters $k = 179, \alpha = 1.5, m = 18000$ for mean packet length 500 bits and $k = 272, \alpha = 1.5, m = 18000$ for mean packet length 750 bits. The time average of queue length in number of packets (Q_N) and in bits (Q_B) and data packet loss probability (P_l) are given in Table 4.2, 4.3 and 4.4 respectively for data packet transmission rate (R_d) of 28800, 14400 and 9600 bps. Note that these parameters are chosen so that the voice outage probability with a ratio of background to interference noise $\eta = 0.1$ is smaller than 0.01.

As can be seen by the results in Tables 4.2 and 4.3, for the same system parameter set and mean data packet length, the packet delay and loss probabilities heavily depend on the packet length distribution. Since the probability of having larger packet sizes is higher for Pareto packet length distribution, the time average queue length has larger values due to the fact that the success probability of larger packets is smaller. Moreover, for the same limitation on the buffer size and retrans-

Parameters	Value
Maximum Buffer Size	25000 bits
Maximum Retransmission Number	10
Voice Mean On Time	1 sec.
Voice Mean Off Time	1.5 sec.
Voice Packet Length	192 bits
Data Packet Interarrival Time	0.1 sec.
Data Packet Retransmission Delay	0.2 sec.
Voice Transmission Rate	9600 bps
Bandwidth	1.25 MHz
Propagation Constant	4

Table 4.1 Parameter Set

mission numbers, the packet loss is much higher for Pareto distributed packet length. For instance, as seen by the numerical results in Table 4.4, it can even be more than hundred times of the packet loss for exponential packet length.

Increasing the transmission rate of common channel, one one hand decreases transmission time of a packet but on the other hand it increases the noise level, therefore increasing the number of retransmission. Table 4.5 displays the results for mean and peakedness (variance/mean) of data packet delay. The average packet delay is the smallest for $R_d = 14400bps$ for this experiment set. For $R_d = 9600bps$, packet transmission time is increased while for $R_d = 28800bps$ the number of retransmissions is increased due to the increased noise level. Note that higher transmission rate causes also degradation of voice calls.

4.3.4 Dynamic Resource Control with Integrated Services

The gain in the CDMA capacity when voice activity detection is used can be achieved by monitoring the voice calls and using the idle periods for the transmission of other voice and data users (Section 1.1). The main goal is to increase the overall capacity and throughput without degrading the QoS seen by the voice users. However even if

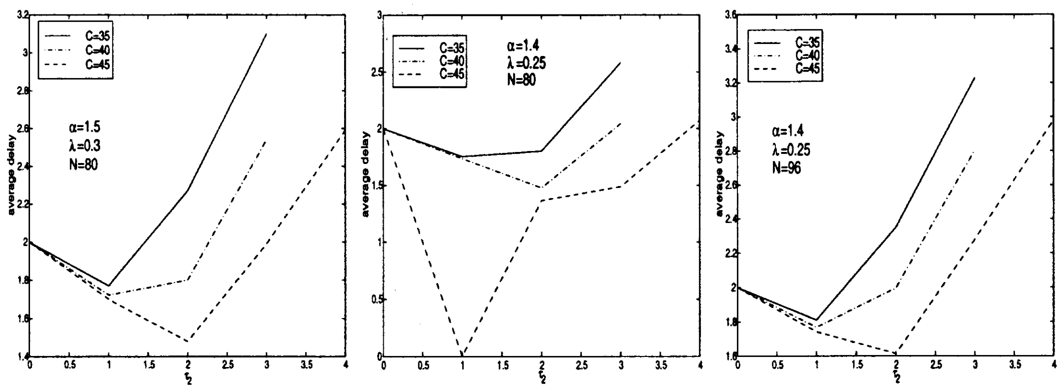


Figure 4.14 Mean packet delay versus timer values

Mean Data Packet Length=500 bits				Mean Data Packet Length=750 bits			
$R_d(bps)$	28800	14400	9600	$R_d(bps)$	28800	14400	9600
fixed	1.19	0.47	0.81	*fixed*	6.28	1.09	2.30
exponential	2.12	0.57	1.09	*exponential*	16.34	1.52	3.82
*Pareto*1	6.12	0.80	1.67	*Pareto*1	18.30	2.17	4.38
*Pareto*2	6.15	0.88	2.02	*Pareto*2	17.24	2.02	3.91

Table 4.2 The time average of queue length in number of packets (Q_N)

Mean Data Packet Length=500 bits			
$R_d(bps)$	28800	14400	9600
fixed	598.49	237.55	404.05
exponential	1300.61	464.66	805.81
*Pareto*1	4190.45	926.38	1611.04
*Pareto*2	4712.88	1120.45	2715.4
Mean Data Packet Length=750 bits			
$R_d(bps)$	28800	14400	9600
fixed	4710.11	819.33	1727.51
exponential	12074.7	1572.63	3447.64
*Pareto*1	14109.2	2605.82	4412.92
*Pareto*2	13563.5	2715.40	4251.87

Table 4.3 The time average of queue length in bits (Q_B)

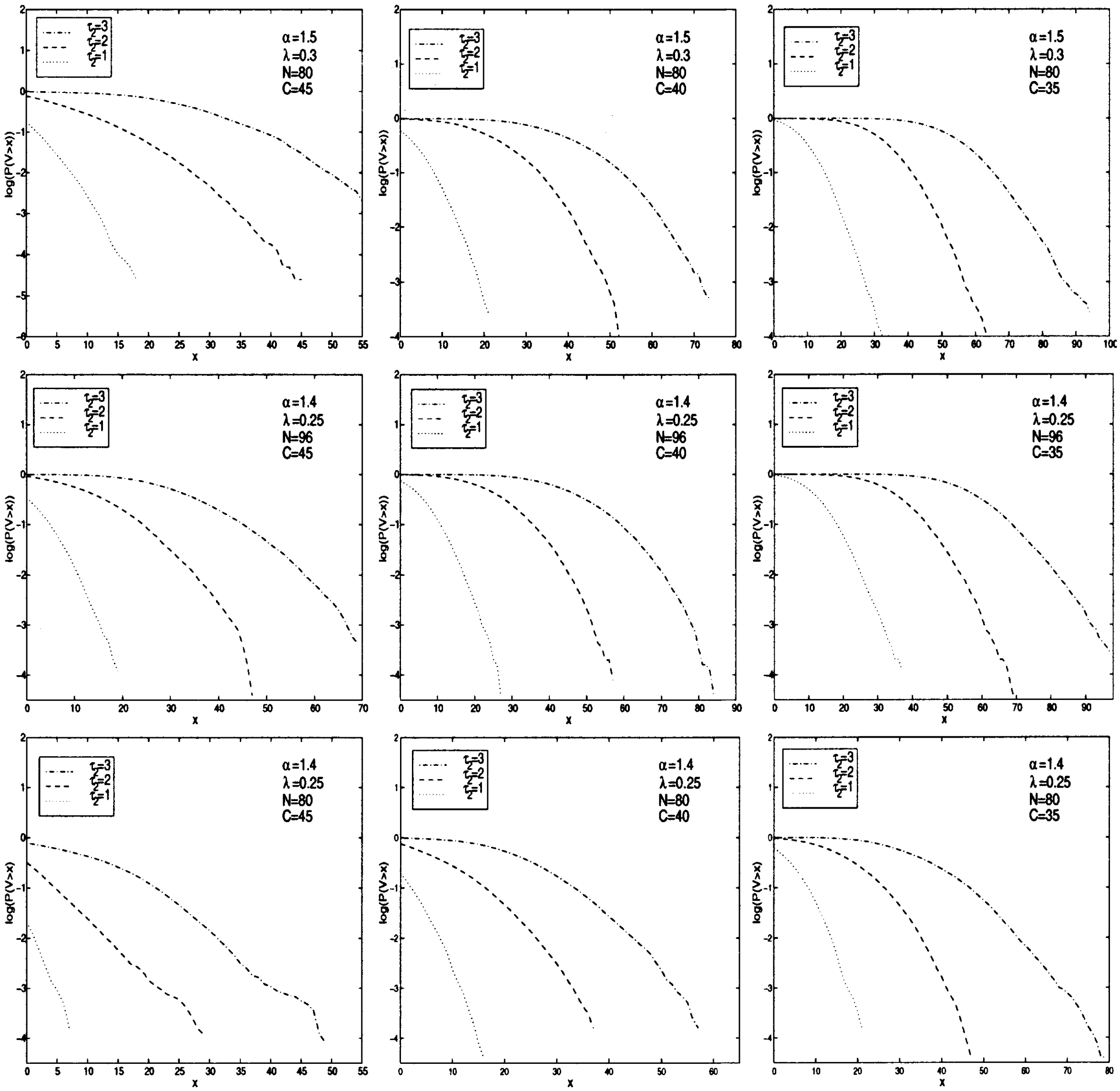


Figure 4.15 Queue length for burst switching, Top: $m = 24, a = 0.46, H = 0.75$, Middle: $m = 24, a = 0.475, H = 0.8$, Bottom: $m = 20, a = 0.6, H = 0.8$

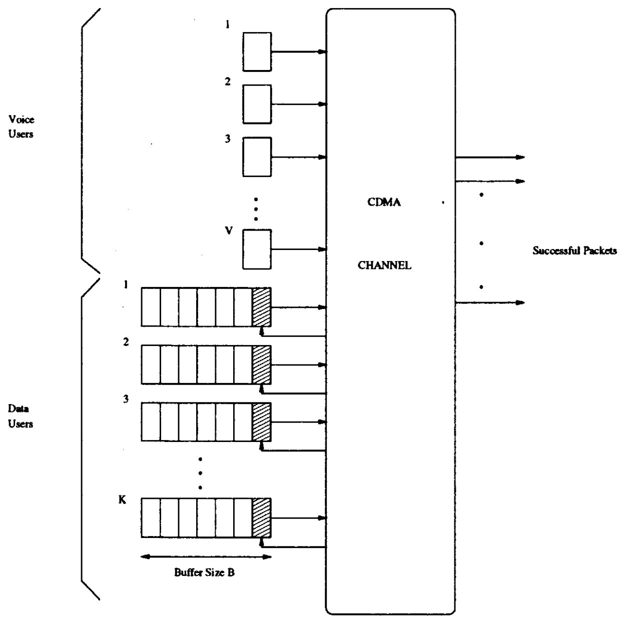


Figure 4.16 System Model

Mean Data Packet Length=500 bits			
$R_d(bps)$	28800	14400	9600
fixed	0	0	0
exponential	0.0003	0	0
*Pareto*1	0.0366	0.0003	0.0004
*Pareto*2	0.0632	0.0025	0.015
Mean Data Packet Length=750 bits			
$R_d(bps)$	28800	14400	9600
fixed	0.0084	0	0
exponential	0.089	0.0001	0.0001
*Pareto*1	0.198	0.0055	0.0063
*Pareto*2	0.199	0.015	0.0176

Table 4.4 Data packet loss probability (P_l)

Mean(Peakedness) Data Packet Length=500 bits			
$R_d(bps)$	28800	14400	9600
fixed	0.109 (0.56)	0.048 (0.04)	0.080 (0.03)
exp.	0.212 (0.89)	0.058 (0.08)	0.109 (0.11)
Pareto1	0.631 (2.67)	0.081 (0.32)	0.166 (0.41)
Pareto2	0.647 (2.85)	0.088 (0.44)	0.167 (0.53)
Mean(Peakedness) Data Packet Length=750 bits			
$R_d(bps)$	28800	14400	9600
fixed	0.635 (0.99)	0.108 (0.13)	0.229 (0.14)
exp.	1.785 (1.53)	0.152 (0.26)	0.38 (0.37)
Pareto1	2.249 (2.02)	0.218 (0.62)	0.442 (0.54)
Pareto2	2.136 (2.22)	0.204 (0.74)	0.398 (0.55)

Table 4.5 Mean and peakedness values of data packet delay

perfect power control (constant E_b/N_o) is assumed, the measurement interval and the data packet duration has impact on the results Capone et al. [12]. These intervals must be chosen so that the probability that a voice user makes a transition to "on state" is small. The data rate $R_d(t)$ at an interval t can be computed from the outage condition:

$$\sum_{i=1}^{V} \nu_i \Gamma_v + \sum_{i=1}^{K} \phi_i \Gamma_d R_d / R_v \leq W/R_v(1-\eta) \tag{4.18}$$

where ν_i and ϕ_i are binary values indicating that the user is on for voice and data user respectively. R_d depends on the number of active voice users at interval t. If the data user has a packet in the queue at that interval, it transmits with rate $R_d(t)$. If a voice user goes on before the transmission of data packets in that interval, the QoS of users will be degraded. To reduce this degradation, the packet is segmented to segments of L_d bits so that average segment transmission ($L_d/E[R_d]$) is 5 msec. The packet delay is computed as the difference between the successful receipt of the last segment of the same packet and the packet generation time.

Mean Data Packet Length=500 bits		
	Mean Packet Delay	Packet Delay Peakedness
fixed	0.088	0.03
exponential	0.116	0.12
*Pareto*1	0.184	0.48
*Pareto*2	0.180	0.61

Table 4.6 Mean and peakedness of packet delay

Mean Data Packet Length=500 bits	
	Q_B
fixed	371.44
exponential	682.48
*Pareto*1	1309.58
*Pareto*2	1328.85

Table 4.7 The time average of queue length in bits (Q_B)

The parameters given in Table 4.1 are used with fixed, exponential, Pareto1 and Pareto2 packet length distributions of mean 500 bits. The effect of different packet length distributions can be seen in the mean and peakedness values for packet delay and the time average of queue length in bits (Q_B) that are displayed in Tables 4.6 and 4.7. The packet loss probability (Table 4.8) is smaller than that of common channel packet transmission. This is due to the fact that the queue is emptied at the segment transmission rate while in common channel access the head of the queue is removed when the whole packet is transmitted successfully. Furthermore, the retransmission probability is almost zero. For the given parameter set, the mean data transmission rate ($E[R_d]$) can be computed as 9330 bps from Eq. 4.18. Since only high priority 10 voice users' activity is tracked, the data transmission rate is computed assuming all 10 data users are active, causing a higher packet delay when compared to the results of common channel packet transmission(Table 4.5).

Mean Data Packet Length=500 bits	
	P_l
$fixed$	0
$exponential$	0
$Pareto1$	0.00004
$Pareto2$	0.00013

Table 4.8 Data packet loss probability (P_l)

The advantage of burst reservation schemes for data services is the minimization of the interference for voice and data packets at the expense of higher overhead to control and measure the channel load. Aloha type common packet transmission requires higher rate of retransmission for data users while a simpler control mechanism is needed. The performance analysis of Aloha type common packet channel transmission is studied in depth in the following chapters.

CHAPTER 5

ANALYSIS OF CDMA-CPCH SYSTEMS WITH GENERAL PACKET LENGTH DISTRIBUTION, FINITE POPULATION AND FINITE BUFFERS

This section covers the analysis of CDMA common packet channel transmission systems for finite population and finite buffers. Most of the related previous work (Section 2.2) is done with the assumptions that there is a infinite number of mobiles or that backlogged users do not generate any new packet. In Okada et al. [26] these assumptions are removed for a fixed packet length. In this section their work is extended for arbitrary packet length in order to analyze the effect of traffic characteristics of multimedia packet data with heavy-tailed packet length distribution since, recently, the heavy-tailed packet length distribution has been widely used in many research and development studies for wireless technologies (Section 3.1). In this section, the analysis is provided for a general packet length distribution in order to compare the performance of the system for fixed, exponential and heavy-tailed packet lengths. Specifically, the system is analyzed in terms of packet delay, packet loss, queue length and throughput. The obtained results demonstrate that for packet length distributions with the same mean but different tail properties the system behavior can change dramatically. For the same average traffic load, packet delay and packet loss are increased while system throughput is decreased for heavy-tailed packet length distributions.

5.1 System Model

Figure 5.1 represents an unslotted random access spread spectrum packet radio network with a finite number of identical users and finite buffer size. Packet inter-arrival time is assumed to be exponentially distributed with mean $1/\lambda$. Fixed, Exponential and Pareto distributed packet lengths are investigated. A new user accesses the system by sending an access message through access channels. The number of access channels is assumed to be high so that access failure is omitted. Then, the user transmits its packets through the assigned channel (code). Note that, only bit synchronization is assumed. The number of interferers during a packet is not constant. Bit error rate probability is computed for each packet segment with constant Signal-to-Noise-Ratio (SNR) value (Section 3.2). Throughout this study, it is assumed that if there is at least one erroneous bit, the packet is assumed to be lost, since most of non-real time data services require probability of error approximately zero. However, error correction can be included in the analysis by changing the corresponding equations according to the number of correctable erroneous bits Joseph et al. [18], (i.e. equations 5.28 and 5.32), without affecting the analytical approach. In the error case, a negative acknowledgment is sent to the corresponding user through control channels. Backlogged users retransmit their packets after an exponentially distributed transmission delay with mean $1/p$. If a packet is in the queue, it is transmitted with the same transmission delay. Therefore, each packet starting a busy period has a different service time distribution than the packets that arrive during an idle period. Packets are stored and served on a first-in-first-out discipline. Each user can have a maximum of B packets including the packet in transmission. A copy of the packet in transmission occupies the top of the queue and it is removed immediately after its successful receipt. Any packet that arrives while the user buffer is full is assumed to be lost.

5.2 System Analysis

Assuming that the system condition changes slowly enough to compute the steady state probabilities, the system is divided into user and channel part [26].

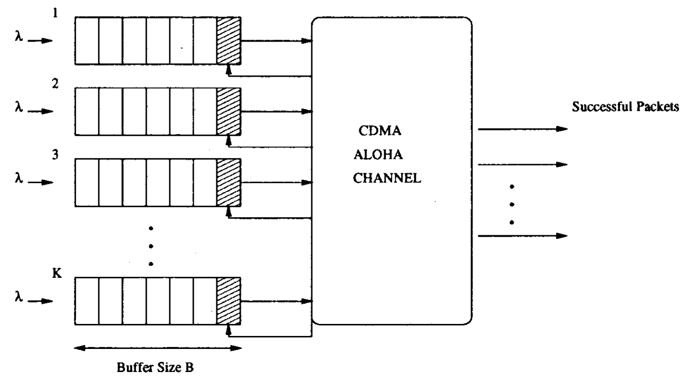


Figure 5.1 System model of CDMA-CPCH systems with finite buffer size

5.2.1 User Part Analysis

Each user is modeled with $M/G/1/B$ queuing model where B is the buffer size which includes the packet in service. Packet delay consists of queuing and service time where service time includes the retransmission times. A packet is retransmitted until its successful receipt. Composite (new + retransmitted) packet duration distribution is denoted by $b(\tau)$ with mean B_c/R where R is the transmission rate and B_c is the mean composite packet length. In the following, it is assumed that the composite packet length follows the same distribution as that of the arrival packet length ($b_a(\tau)$ with mean A/R).

A user in idle mode transmits the new packet with rate λ while it retransmits the unsuccessful packet and the stored packet with rate p. The pdf of the service time of a packet of duration τ for the user who is initially in the idle mode is:

$$d_I(t,\tau) = Q_s(\tau)\delta(t-\tau) + \sum_{m=1}^{\infty}(1-Q_s(\tau))^m Q_s(\tau)E_m(t-(m+1)\tau;p/m) \quad (5.1)$$

where $\delta(t)$ is a delta function, $E_k(t;v)$ is the pdf of k-Erlangian distribution with an average $1/v$, Q_s denotes the packet success probability and is dependent on the packet length, number of interferers and background noise.

The pdf of the service time of a packet of duration τ for the user who is initially in the busy mode is:

$$d_B(t,\tau) = \sum_{m=1}^{\infty}(1-Q_s(\tau))^{m-1}Q_s(\tau)E_m(t-m\tau;p/m) \tag{5.2}$$

The analytical model for an $M/G/1/B$ queue is computed to find the steady state probability of having j packets at the user station (Kleinrock et al. [65], Cooper et al. [66]) in order to compute system characteristics such as average queue length, delay and rejection probability.

The system is considered a $(B-1)$-state Markov chain where the state transition occurs at successful packet transmissions. The probability p_{Ij} (p_{Bj}) that j packets are arriving at the user station which is initially in idle (busy) mode during the service is computed from the Laplace-Stieljes transform of service time distributions:

$$p_{Ij} = \int_{\tau=0}^{\infty}\frac{1}{j!}\lim_{z\to 0}\frac{\delta^j}{\delta z^j}L_I((1-z)\lambda,\tau)\ b_a(\tau)\,d\tau \tag{5.3}$$

$$p_{Bj} = \int_{\tau=0}^{\infty}\frac{1}{j!}\lim_{z\to 0}\frac{\delta^j}{\delta z^j}L_B((1-z)\lambda,\tau)b_a(\tau)\,d\tau \tag{5.4}$$

where Laplace-Stieljes transform of $d_I(t,\tau)$ and $d_B(t,\tau)$ are computed as:

$$L_I(s,\tau) = \sum_{m=0}^{\infty}(1-Q_s(\tau))^m Q_s(\tau)exp(-(m+1)\tau s)(\frac{p}{s+p})^m \tag{5.5}$$

$$L_B(s,\tau) = \sum_{m=1}^{\infty}(1-Q_s(\tau))^{m-1}Q_s(\tau)exp(-m\tau s)(\frac{p}{s+p})^m \tag{5.6}$$

The average service times can be computed by averaging $d_I(t,\tau)$ and $d_B(t,\tau)$ or by differentiating their Laplace-Stieljes transforms:

$$1/\mu_I = \int_{\tau=0}^{\infty} (\tau + \frac{\tau + 1/p}{Q_s(\tau)}(1 - Q_s(\tau)))\ b_a(\tau)\, d\tau \tag{5.7}$$

$$1/\mu_B = \int_{\tau=0}^{\infty} (\frac{\tau + 1/p}{Q_s(\tau)})\ b_a(\tau)\, d\tau \tag{5.8}$$

Let q_{Ij} and q_{Bj} denote the limiting probability distributions of having j packets left behind by the successful packet at idle and busy user stations, respectively. The probability of having j packets at a certain user station immediately after completion of a packet transmission ($\Pi_j = q_{Ij} + q_{Bj}$) is derived as the asymptotical equation:

$$\Pi_{j+1} = (\Pi_j - p_{Ij}\Pi_0 - \sum_{m=1}^{j} p_{Bj-m+1}\Pi_m)(p_{B0})^{-1} \quad j = 0, 1, ...B-2 \tag{5.9}$$

$$\sum_{i=0}^{B-1} \Pi_j = 1 \tag{5.10}$$

The steady state probabilities (P_j) can be computed by defining $C_j = \Pi_j/\Pi_0$:

$$P_j = \frac{\Pi_j}{\Pi_0 + a} = \frac{C_j}{1 + aC} \quad j = 0, 1, ..., B-1 \tag{5.11}$$

$$P_B = 1 - \frac{1}{\Pi_0 + a} = 1 - \frac{C}{1 + aC} \tag{5.12}$$

where a is the traffic intensity at the user station:

$$a = \lambda(\Pi_0/\mu_I + (1 - \Pi_0)/\mu_B) \tag{5.13}$$

$$aC = \lambda/\mu_I + \lambda/\mu_B(C - 1) \tag{5.14}$$

$$C = 1 + \sum_{i=1}^{B-1} C_j \tag{5.15}$$

P_B is the blocking probability and P_0 is the ratio of the number of the idle user stations to the number of all users.

The mean (D) and the standard deviation (σ_D) of packet delay are computed by differentiating the Laplace-Stieljes transform of packet waiting time distribution $D(s)$ Courtois et al. [76]:

$$
\begin{aligned}
D &= -\frac{d}{ds}D(s)|_{s=0} && (5.16)\\
\sigma_D^2 &= \frac{d^2}{ds^2}D(s)|_{s=0} - D^2 && (5.17)
\end{aligned}
$$

where

$$
\begin{aligned}
D(s) &= \frac{1}{(1-Qr)\sigma\lambda}\sum_{j=1}^{B-1}(L_I(s)(q_{I0}+q_{B0})\\
&\quad (\frac{\lambda}{\lambda-s})^j + L_B(s)\sum_{i=1}^{j}(q_{Ii}+q_{Bi})\\
&\quad (\frac{\lambda}{\lambda-s})^{j-i+1} - \sum_{i=0}^{j-1}(q_{Ii}+q_{Bi})\\
&\quad (\frac{\lambda}{\lambda-s})^{j-i})(L_B(s))^j && (5.18)
\end{aligned}
$$

where

$$
\sigma = (q_{I0}+q_{B0})(\frac{1}{\mu_I} - \frac{1}{\mu_B} + \frac{1}{\lambda}) + \frac{1}{\mu_B} \qquad (5.19)
$$

Note that average packet delay is equal to the sum of the average service time s and average waiting time w of data packets:

$$
\begin{aligned}
s &= \frac{1}{\mu_I}\Pi_0 + \frac{1}{\mu_B}(1-\Pi_0) && (5.20)\\
w &= D - s && (5.21)
\end{aligned}
$$

Average number (a_r) of transmissions per packet can be computed as:

$$a_r = \int_0^{\infty}(Q_s(\tau)\sum_{i=0}^{\infty}(i+1)(1-Q_s(\tau))^i)b_a(\tau)d\tau = \int_0^{\infty} 1/Q_s(\tau)b_a(\tau)d\tau \quad (5.22)$$

5.2.2 Channel Part Analysis

For a finite population of K users, the channel part can be modeled as $M/G/\infty//K$ queue model which is invariant to the service time distribution (Cooper et al. [66]).

The average number of data packets G_{sys} transmitted to the channel during a data packet duration and normalized average number of data packets G'_{sys} transmitted in 1 second are given by:

$$G_{sys} = \frac{K\lambda_c/\mu_c}{1+\lambda_c/\mu_c} \quad (5.23)$$

$$G'_{sys} = G_{sys}\,\mu_c \quad (5.24)$$

where $\mu_c = R/B_c$ and λ_c is actual birth rate.

The number of packets transmitted to the channel per unit time is equal to the number of total transmissions of packets that gained service (that are accepted to the terminal queue):

$$G'_{sys} = K\lambda a_r(1-P_B) \quad (5.25)$$

The birth rate (λ_c) for actually transmitted packets can then be computed from Eq. 5.23-5.24.

The probability that the packet is transmitted successfully from the first bit to the $(i-1)^{th}$ bit with k interfering packets on the i^{th} bit is $P_s(k,i)$:

$$P_s(k,1) = \frac{(\lambda_c/\mu_c)^k \frac{(K-1)!}{k!(K-k-1)!}}{(1+\lambda_c/\mu_c)^{K-1}} \text{ if } k < K \quad (5.26)$$

$$P_s(k,1) = 0 \text{ otherwise} \tag{5.27}$$

For $i > 1$:

$$\begin{aligned} P_s(k,i) &= P_s(k,i-1)(1 - k\mu_c\Delta t - (K-1-k)\lambda_c\Delta_t) \\ &\quad (1 - P_b(k)) + P_s(k+1,i-1)(k+1) \\ &\quad \mu_c\Delta t(1 - P_b(k+1)) + P_s(k-1,i-1) \\ &\quad (K-k)\lambda_c\Delta_t(1 - P_b(k-1)) \end{aligned} \tag{5.28}$$

It is assumed that, the channel load is constant during one bit duration (Δt) and that the number of interfering packets can change at most by 1 between the consecutive bits. In the next section, it is shown via simulation that this assumption is valid for different arrival processes given that transmission rate is high enough.

The bit error rate ($P_b(k)$) is expressed as Holtzman [64]:

$$\begin{aligned} P_b(k) &= 2/3Q[(\frac{k}{3G} + \frac{No}{2Eb})^{-0.5}] + 1/6Q[(\frac{kG/3 + \sqrt{3}\sigma}{G^2} + \frac{No}{2Eb})^{-0.5}] \\ &\quad + 1/6Q[(\frac{kG/3 - \sqrt{3}\sigma}{G^2} + \frac{No}{2Eb})^{-0.5}] \\ \sigma^2 &= k(G^2\frac{23}{360} + G(\frac{1}{20} + \frac{k-1}{36}) - \frac{1}{20} - \frac{k-1}{36}) \end{aligned} \tag{5.29}$$

where G is the number of chips per bit, k is the number of interfering packets, E_b is the bit energy of the signal, $No/2$ is two-sided spectral density of AWGN.

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty exp(-u^2/2)du \tag{5.30}$$

The throughput S in a packet duration is computed using packet success probability Qs:

$$S = \int_{\tau=0}^{\infty} G_{sys} \sum_{k=0}^{\infty} P_s(k, L(\tau))(1 - P_b(k))\ b(\tau)\, d\tau \tag{5.31}$$

$$Qs(\tau) = \sum_{k=0}^{\infty} P_s(k, L(\tau))(1 - P_b(k)) \tag{5.32}$$

where $L(\tau) = \tau R$. The mean number of successful packets in 1 second is then computed as $S' = S\,\mu_c$.

5.2.3 Combination of the Parts

A search loop is formed for λ_c and mean composite packet length (B_c) to compute the throughput using channel part model. Then using the equilibrium equation, rejection probability (Q_r) is computed as:

$$Q_r = 1 - S'/(K\lambda) \tag{5.33}$$

The average delay (D), average queue length (Q) and rejection probability are calculated at the user part as:

$$Q = \sum_{j=1}^{B} jP_j \tag{5.34}$$

$$D = Q/S'/K \tag{5.35}$$

$$Q_r = P_B \tag{5.36}$$

The search of steady-state values is completed when λ_c in both parts is the same and the equilibrium condition is satisfied. If the system has more than one solution, the system exhibits the bistable behavior and the performance curves change discontinuously.

5.3 Numerical Study and Evaluation

This section presents numerical results obtained from the system analysis and the simulation results obtained by using Opnet Modeler. The aim of the study performed here is two-fold. First to validate the assumptions used in the analysis and second to demonstrate the effect of packet length distribution on the system characteristics.

5.3.1 Simulation Model

Before proceeding with presentation of the results, the simulation structure is described briefly. The model consists of one base station that serves K mobile users distributed randomly and uniformly in a hexagonal cell (Figure 5.2). Each mobile accepted to the system is assigned a transmission code. The transmission rate is fixed and predetermined. The transmit power is adapted by a control message from the base station whenever there is a change in the number of admitted users. The user sends its packet immediately if its buffer is empty, otherwise it inserts its packet at the tail of the queue if the buffer size is not exceeded. The packet in transmission is removed from the queue when the corresponding positive acknowledgment message from the base station is received. The base station computes the interference from other users' packets and the background noise to obtain the SNR for each segment of the packet. The simulations were run for a sufficiently long period of time to collect statistics after the system reaches its steady state.

5.3.2 Numerical Results And Discussion

In this section, the numerical results based on the analysis provided in Section 5.2 and the simulation model described in the previous subsection are presented. Two sets of experiments were designed for this purpose. The first one is used to validate the assumption that the channel load is constant during one bit duration and that the number of interfering packets can change at most by 1 between consecutive bits. The second set is used to validate the assumption that the composite and successful packet duration have the same distribution as that of arrival packet duration and study the impact of packet length distribution on the system performance.

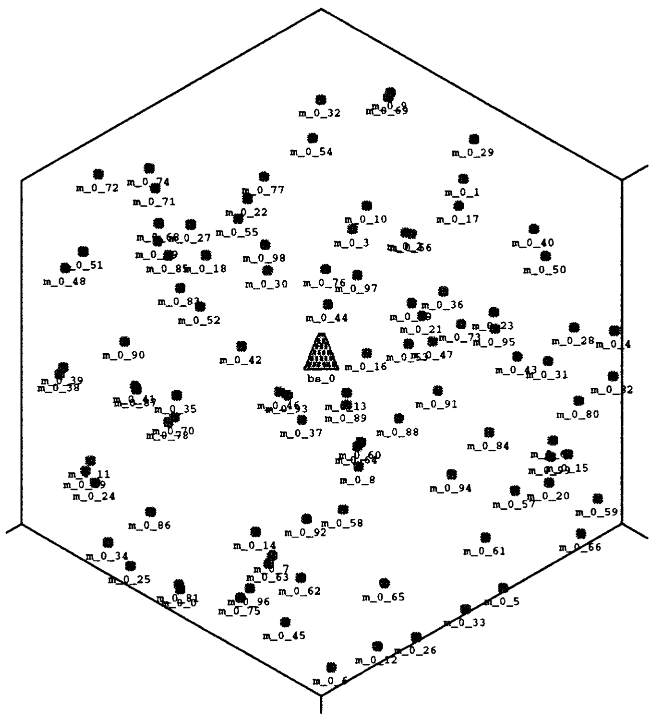


Figure 5.2 Simulation Model.

In order to validate the first assumption that the channel load is constant during one bit duration and that the number of interfering packets can change at most by 1 between consecutive bits for systems with high transmission rate even if the arrival process is not Poisson, Packet interarrival times are Pareto distributed with rate of 1.67 packets per second ($k = 0.2, \alpha = 1.5$) and $W = 1.25$ MHz, $R = 9600$ bps, $E_b/N_0 = 10$ dB. The probability of having m_{i+1} interferers at the $(i+1)^{th}$ bit when there is m_i interferes at the $(i)^{th}$ bit ($p(m_{i+1}|m_i)$) are displayed in Figure 5.3 for packet length of 128 bits. The horizontal axis denotes the number of interferers at the i^{th} bit. The vertical axis denotes the conditional probability of having one less interferer, one more interferer and the same number of interferers at $(i+1)^{th}$ bit from top to bottom respectively. The analytical results are obtained from the Markovian process used in the computation of packet success probability while the simulation results are obtained by counting the number of interfering packets during the transmission of each bit of the tagged packet. As can be seen from Figure 5.3, simulation and analytical results agree and therefore, demonstrate that the number of interfering packets change at most by one between the consecutive bits. The probability $p(m_1)$ of having m_1 interferers for the first bit is displayed in Figure 5.4. The circle represents the probability values computed via simulation by counting other users' packets that are transmitted simultaneously during the transmission of the first bit of the tagged packet while asterisk represents the probability values computed analytically by using (Markovian Model) Poisson distribution with mean $\lambda L/R$.

In order to validate the assumption that the composite and successful packet duration have the same distribution as that of arrival packet duration and study the impact of packet length distribution on the system performance, the following simulation parameter set is used: $K = 100$ (number of users), $\lambda = 0.1$ (exponential packet arrival rate per user), $A = 500bits$ (mean arrival packet length), $E_b/N_0 = 10dB$, $G = 60$ (processing gain), $\gamma = 4$ (path loss factor), $R = 500bits/sec$ (data rate), $t_f = 30MHz$ (base frequency), $W = 30kHz$ (bandwidth). The retransmission

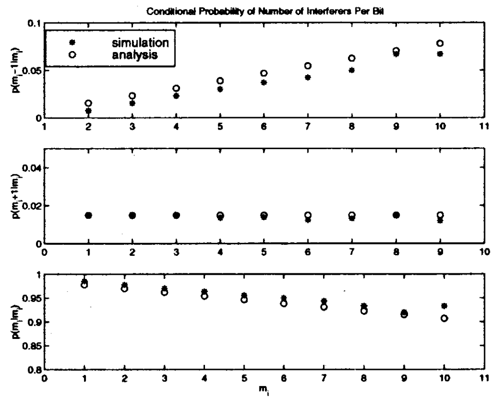
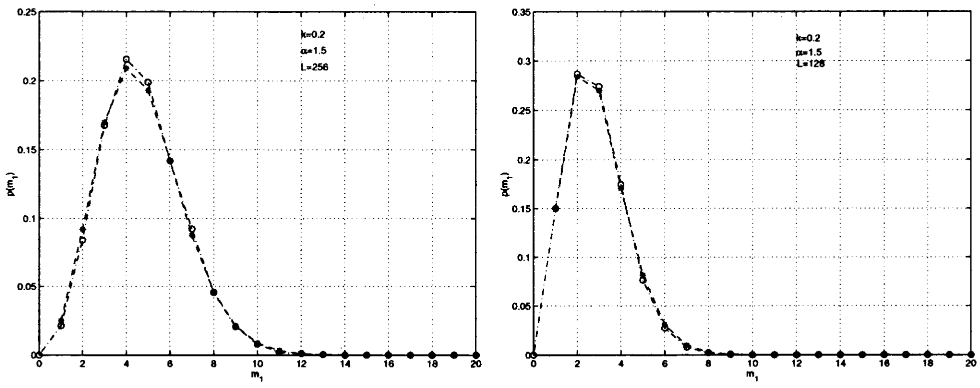


Figure 5.3 Conditional probabilities of number of interferers per bit.

rate p is chosen so that the system is stable for the chosen buffer sizes. The graph on the top-left of Figure 5.5 displays the simulation results for the distribution of composite and successful packet size when the arrival packet length distribution is exponential. It can be seen that the composite and successful packet length distribution conform to exponential distributions with different mean values displayed on the top-right of the figure. Since the probability of success is less for packet of longer duration, the composite mean packet length is higher than that of arrival packet length for variable packet length distributions.

The same observations hold true when arrival packet length distribution is Pareto with cutoff (bottom of the Figure 5.5). Since the minimum (k) and maximum (cutoff) values of composite and successful packet lengths are the same with those of arrival packet length, α (degree of tail heaviness) changes between arrival, successful and composite packet length distributions; it is heavier for composite packet length depending on the system load and traffic characteristics. The simulation results displayed on the bottom-right of the figure conform to Pareto distributions with different α values.



(a) (b)

Figure 5.4 Probability of number of interferers for the first bit for packet length a) 256 bits and b) 128 bits.

In the following, the effect of packet length distribution on the system model is demonstrated by using the second simulation set. Tables 5.1,5.2,5.3,5.4 present the mean (D) and standard deviation (σ_D) of packet delay, average queue length (Q) at each user's buffer, packet rejection probability (Q_r), the normalized throughput per second (S') and the normalized input per second (G'_{sys}) to the channel for fixed, exponential and Pareto packet length distributions respectively. As can be seen from the results displayed in Tables 5.1,5.2,5.3,5.4 the theoretical and simulation results agree. For the same average packet length, the best performance is obtained for fixed packet length distribution. The retransmission delay must be longer to provide the stability for the Pareto packet length distribution due to the higher probability of having longer packets (Tables 5.3 and 5.4). When the cutoff parameter is low, i.e., the tail of the distribution is lighter, the difference between the results (Tables 5.2 and 5.3) is small. However, when the cutoff parameter is large (Table 5.4), a severe degradation of the system performance is observed. The effect of cutoff parameter on packet delay and normalized throughput can be seen for different buffer sizes.

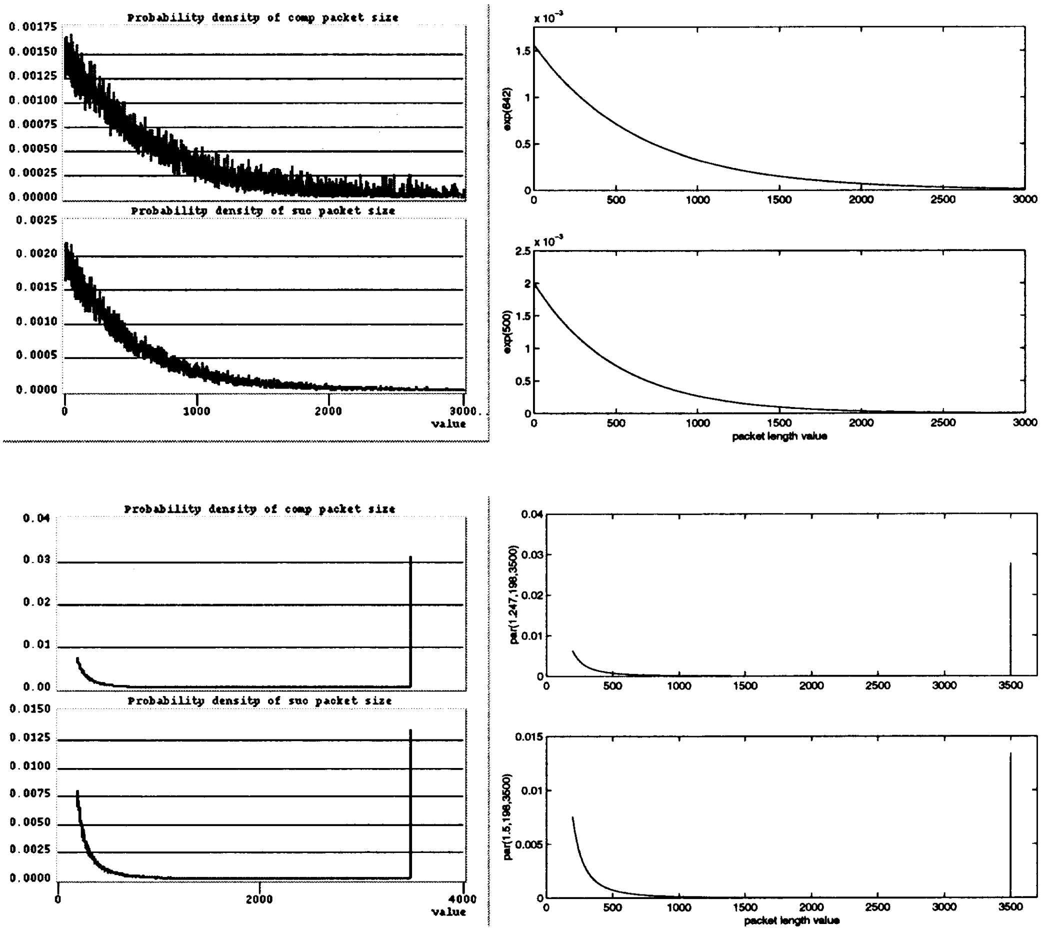


Figure 5.5 Top Left: Simulation results for composite and successful packet length distribution for Exponential arrival packet length ($p = 1/20, A = 500, B_c = 642, B = 3$); Top Right: Exponential distributions with mean 500 and 642; Bottom Left: Simulation results for composite and successful packet length distribution for Pareto arrival packet length ($p = 1/25, A = 500$ with $Pareto(k = 198, \alpha = 1.5, cutoff = 3500), Bc = 605$ with $Pareto(k = 198, \alpha = 1.247, cutoff = 3500), B = 3$); Bottom Right: Pareto distributions with parameters $k = 198, cutoff = 3500, \alpha = 1.5$ and $\alpha = 1.247$.

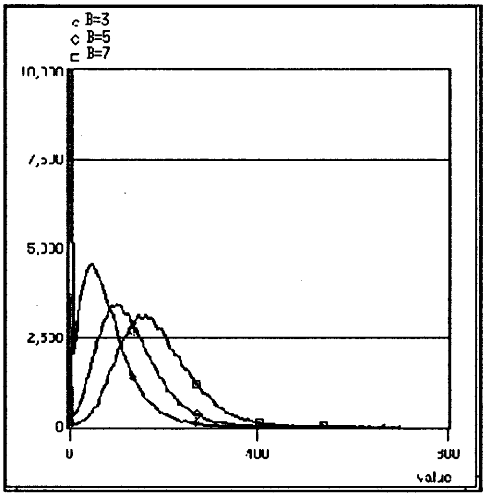


Figure 5.6 Histogram of packet delays for Pareto($k = 199, \alpha = 1.5, cutoff = 3500$), $p = 1/25$.

	B=3		B=5		B=7	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
D	35.38	35.95	68.58	69.51	102.85	104.7
σ_D	30.9	29.79	41.57	41.38	49.64	49.06
Q	1.99	2.00	3.85	3.86	5.77	5.82
Q_r	0.44	0.443	0.439	0.443	0.438	0.443
S'	5.58	5.56	5.58	5.56	5.58	5.56
G'_{sys}	9.07	9.09	9.12	9.09	9.11	9.09

Table 5.1 Results for fixed packet length ($p = 1/10$)

	B=3		B=5		B=7	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
D	58.2	58.98	110.5	110.2	163.5	162.63
σ_D	58.07	54.5	65.47	63.65	73.47	71.7
Q	2.33	2.328	4.36	4.33	6.35	6.36
Q_r	0.607	0.605	0.605	0.606	0.601	0.609
S'	3.97	3.94	3.95	3.93	3.92	3.91
G'_{sys}	5.18	5.15	4.83	4.81	4.76	4.73

Table 5.2 Results for Exponential packet length ($p = 1/20$)

	B=3		B=5		B=7	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
D	64.2	63.59	126.7	126.54	187.2	187.96
σ_D	61.4	54.88	69.91	67.01	79.20	75.8
Q	2.4	2.368	4.44	4.44	6.49	6.48
Q_r	0.63	0.63	0.65	0.651	0.652	0.657
S'	3.7	3.72	3.5	3.51	3.47	3.45
G'_{sys}	4.36	4.366	3.92	3.91	3.84	3.836

Table 5.3 Results for Pareto packet length ($p = 1/25, Pareto(k = 198, \alpha = 1.5, cutoff = 3500)$)

	B=3		B=5		B=7	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
D	95.02	95.64	198.13	198.61	285.56	285.97
σ_D	83.73	81.74	98.78	94.5	113.9	115.47
Q	2.60	2.57	4.693	4.688	6.694	6.70
Q_r	0.729	0.73	0.765	0.764	0.765	0.766
S'	2.72	2.69	2.36	2.36	2.345	2.343
G'_{sys}	2.90	2.86	2.47	2.46	2.44	2.434

Table 5.4 Results for Pareto packet length $p = 1/40, Pareto(k = 186, \alpha = 1.5, cutoff = 8000)$)

The results presented in this section differ from the results of the previously analyzed systems for CDMA random access. For instance, the results presented here show that even if the average traffic load is the same, the packet delay increases significantly when the packet length distribution has heavier tail. This is in contrast to the systems where new message generation is inhibited during a backlog mode and therefore use of the relatively long retransmission delay required for stability does not involve significant average delay penalty, except at very high load Joseph et al. [17]. The results presented in this section demonstrate that the specific packet length characteristics significantly impact the system performance as opposed to the results of random access CDMA systems with simpler assumptions where the results for different length packets generally agree indicating that the results are not strongly dependent on the packet length distribution Storey et al. [25].

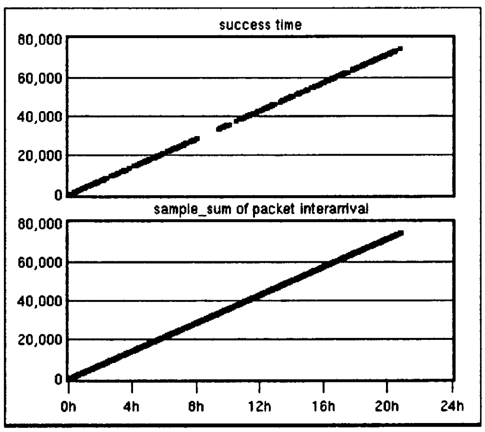


Figure 5.7 Packet receipt times (top) for the user with exponential packet generation times (bottom); vertical axis denotes packet indices at the receiver (top) and at the transmitter (bottom) and horizontal axis denotes the observation time.

Furthermore, the analysis can be used for system optimization. For instance, the optimum buffer size can be computed for the given system parameters and user characteristics. The smoothing effect of buffer size (B) can be seen in Figure 5.6 where the peakedness (σ_D^2/D) is smaller when buffer size is increased. When the cutoff parameter of the heavy tailed distribution is high, i.e., the peakedness of the packet length distribution is high, unacceptable packet delays are observed due to the retransmission of the same very long packet. For instance, Figure 5.7 illustrates such a scenario. The horizontal axis denotes the simulation time while the vertical axis denotes the indices of packets that are successfully transmitted (Figure 5.7,Top) for packets arrived according to an exponential interarrival time (Figure 5.7, Bottom). The big gap observed in this figure between successful transmission times is due to the retransmission of the same packet. Such extreme cases are not observed for fixed or light tailed packet length distributions. In the next section, the system is analyzed by restricting the time a packet can spend in the system, specifically by limiting the waiting and serving time of a packet.

CHAPTER 6

ANALYSIS OF CDMA-CPCH SYSTEMS WITH GENERAL PACKET LENGTH DISTRIBUTION, FINITE POPULATION AND FINITE SOJOURN TIME

In CPCH-based transfer schemes (Section 3.3.4), a discard function allows to discharge the data unit from the buffer on the transmitter side, when the transmission of the data does not succeed for a long time[67]. There are several alternative operation modes of the discard function, and which discard function to use will be mainly driven by the QoS requirements of the Radio Access Bearer. One of the possibilities is to discard the data unit after a maximum number of retransmissions [68]. This alternative uses the number of retransmissions as a trigger for data discard, and is therefore only applicable for acknowledged mode RLC. It makes the data discard function dependent of the channel rate. Also, this variant of the data discard function strives to keep the data loss rate constant for the connection, on the cost of a variable delay. The described mechanism can be analyzed as an asynchronous CDMA Aloha system with general packet length distribution, finite population and finite sojourn time.

Figure 2.1 displays the system model for a CDMA scheme. Each user starts transmitting after a transmission code is assigned. Due to the multiuser interference and background noise, some of the packets will be received with errors. In this case, the user must retransmit the same packet after a random delay. Hence, the channel input consists of new packets and retransmitted packets.

This chapter extends the analysis of CDMA-CPCH systems studied in Chapter 5 where the assumptions that are used in the literature (Section 2.2) are removed and the system behavior is analyzed for the more general and realistic case with finite population, finite sojourn time and variable packet length. Since the system performance is investigated for variable packet length distribution, the system is studied in terms of unfinished workload. The packet sojourn time is restricted by the quality of service requirements of packets in terms of delay parameters. The sojourn time is defined as the time spent in the system (queue+service) by the tagged packet. In this chapter, the analysis is provided for a general packet length distribution and the performance of the system is compared for fixed, exponential and heavy-tailed packet lengths. Specifically, the system is analyzed in terms of packet delay, packet loss and throughput.

Furthermore, the analysis can be used for optimum resource allocation and call admission. Given the system parameters such as bandwidth, background noise, path loss factor, and user requirements such as loss and delay bounds, the proposed analytical method provides an accurate and efficient tool that can be used to adjust the system variables e.g., channel rate, maximum packet length to be transmitted, maximum number of users that can be accepted in order to increase the system performance. Note that, the analysis is done with regard to both completely served and lost packets (packets can be lost either after waiting some limited prespecified time in the buffer or after reaching a maximum number of retransmissions). Therefore, it can be used to quantify the total system utilization such as the buffer time occupied by the packets lost due to the impatience factor and the service time of packets lost due to the limit on the number of retransmissions.

6.1 System Model

Figure 6.1 represents an unslotted random access spread spectrum packet radio network with a finite number (K) of identical users. Packet interarrival time is assumed to be exponentially distributed with mean $1/\lambda$. Packet lengths may follow an arbitrary (general) probability distribution. A new user accesses the system by sending an access message through the access channels. The number of access channels is assumed to be high enough so that access failure is omitted. Then, the user transmits its packets through the assigned channel (code). Note that, only bit synchronization is assumed as in Chapter 5. The number of interferers during a packet is not constant. Bit error rate probability is computed for each packet segment with constant Signal-to-Noise-Ratio (SNR) value.

Throughout this study, it is assumed that if there is at least one erroneous bit the packet is lost, since most of non-real time data services require probability of error approximately zero. However, error correction can be included in the analysis by changing the corresponding equations according to the number of correctable erroneous bits Joseph et al. [18] without affecting the analytical approach as stated in the previous section. In the error case, a negative acknowledgment is sent to the corresponding user through control channels. Backlogged terminals retransmit their packets after an exponentially distributed transmission delay with mean $1/p$. If a packet is in the queue, it is transmitted with the same transmission delay. Therefore, each packet starting a busy period has a different service time distribution than the packets that arrive during an idle period of the terminal. Packets are stored until a predefined waiting time and are served on a first-in-first-out discipline. A copy of the packet in transmission occupies the top of the queue and it is removed immediately after its successful receipt or if it reaches the limit on the number of retransmissions without being successfully transmitted. Hence, a packet may be dropped due to the following reasons: either if it waits longer than a predefined time or if it has been unsuccessful after a maximum number of retransmissions.

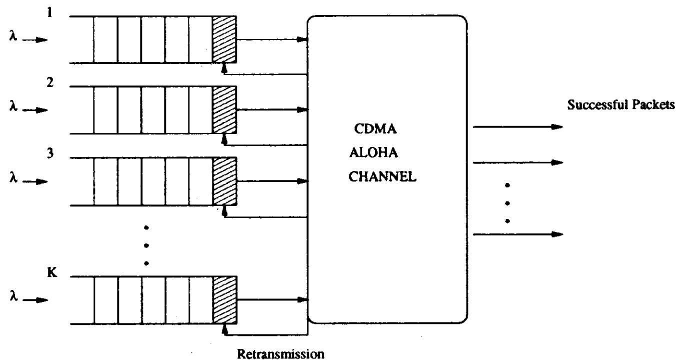


Figure 6.1 System model of CDMA-CPCH systems with finite sojourn time

6.2 System Analysis

As stated in Chapter 5, assuming that the system condition changes slowly enough to compute the steady state probabilities, the system can be divided into user and channel part.

6.2.1 User Part Analysis

Each user is modeled with $M/G/1$ queuing model with bounded sojourn time. Packet delay consists of queuing and service times where service time includes the retransmission times. Composite (new + retransmitted) packet duration distribution is denoted by $b(\tau)$ with mean B_c/R where R is the transmission rate and B_c is the mean composite packet length. In the following, it is assumed that the composite packet length follows the same distribution as that of the arrival packet length ($b_a(\tau)$ with mean A/R). This assumption is shown to be valid via simulation in the next section. Packets wait for service only for a limited time (τ_m) and are retransmitted for a limited number (T) of retransmissions. Any packet that has waited in the buffer for a fixed time τ_m without having started its service, or has been unsuccessful after a maximum number of retransmissions T is assumed to be lost. Hence, the total sojourn time is limited by a variable time dependent on the channel rate [68]. Note that, the queue of the considered system is not emptied at a constant rate since the service rate depends on the packet success probability. For this reason, the workload is expressed in terms of time units.

A terminal in idle mode transmits the new packet with rate λ while it retransmits the unsuccessful packet and the stored packet with rate p. Therefore, a packet of length τ that is generated by an idle terminal receives exceptional service with distribution function $D_I(t,\tau)$ while it is assigned service time of distribution function $D_B(t,\tau)$ if it is generated by a busy terminal. The density function of the service time of a packet of duration τ generated by a terminal that is initially in the idle mode is:

$$\begin{aligned} d_I(t,\tau) &= Q_s(\tau)\delta(t-\tau) + \sum_{m=1}^{T}(1-Q_s(\tau))^m Q_s(\tau) E_m(t-(m+1)\tau; p/m) \\ &\quad + (1-Q_s(\tau))^{T+1} E_T(t-(T+1)\tau; p/T) \end{aligned} \tag{6.1}$$

where $\delta(t)$ is a delta function, $E_k(t;v)$ is the pdf of k-Erlangian distribution with an average $1/v$, T is the maximum number of retransmissions, Q_s denotes the packet success probability and is dependent on the packet length, number of interferers and background noise.

The probability density function of the service time of a packet of duration τ generated by a terminal that is initially in the busy mode is:

$$\begin{aligned} d_B(t,\tau) &= \sum_{m=1}^{T+1}(1-Q_s(\tau))^{m-1} Q_s(\tau) E_m(t-m\tau; p/m) \\ &\quad + (1-Q_s(\tau))^{T+1} E_{T+1}(t-(T+1)\tau; p/(T+1)) \end{aligned} \tag{6.2}$$

The parameters of interest are the long-run fraction of packets that are lost during waiting time and service time, the long-run average delay of a served and lost packet and the system throughput. The $M/G/1$ queue with restricted waiting time can be analyzed using a modified system where a packet is rejected immediately if upon arrival the amount of work in the system is greater than some threshold Tijms et al. [73]. The admitted packets in the modified system are exactly the served packets in the original system. Using the property "Poisson arrivals see time averages" [73], the parameters of interest of the original system can be expressed in terms of the limiting distribution function $V_{\tau_m}(x)$ of the amount of work in the modified system. The long-run fraction (π_w) of packets that are lost during waiting time, the long-run fraction (π_r) of served packets that are lost due to the restricted number of transmissions, the long-run average waiting time (w_q) of a served and lost packet, the long-run average waiting time (w_s) of a served packet and average number (a_r) of transmissions can be computed as:

$$\pi_w = 1 - V_{\tau_m}(\tau_m) \tag{6.3}$$

$$\pi_r = \int_0^\infty (1 - Q_s(\tau))^{T+1} b(\tau) d\tau \tag{6.4}$$

$$w_q = \tau_m \ \pi_w + \int_0^{\tau_m} x \ v_{\tau_m}(x) \ dx \tag{6.5}$$

$$w_s = \frac{w_q - \tau_m \pi_w}{1 - \pi_w} \tag{6.6}$$

$$\begin{aligned} a_r &= \int_0^\infty (Q_s(\tau) \sum_{i=0}^{T} (i+1)(1 - Q_s(\tau))^i + (1 - Q_s(\tau))^{T+1}(T+1)) b(\tau) d\tau \\ &= \int_0^\infty \frac{1 - (1 - Q_s(\tau))^{T+1}}{Q_s(\tau)} b(\tau) d\tau \end{aligned} \tag{6.7}$$

$V_{\tau_m}(x)$ has a positive mass at $x = 0$ and a density $v_{\tau_m}(x)$ for $x > 0$. It is possible to express $V_{\tau_m}(x)$ in terms of $V_\infty(x)$ which is the limiting distribution function of the amount of work in the modified system for exceptional service:

$$V_{\tau_m}(x) = c \ V_\infty(x) \ \ 0 \leq x \leq \tau_m \tag{6.8}$$

where c is a positive constant. To find the constant c, the equality between the long-run average number of downcrossings of level x of the workload process per unit time and the long-run average number of upcrossings of level x of the workload per unit time is used. This equality yields the following integral equation:

$$\begin{aligned} v_{\tau_m}(x) &= \lambda(1 - L_I(x))V_{\tau_m}(0) \\ &+\lambda \int_0^{min(x,\tau_m)} (1 - L_B(x-y))\ v_{\tau_m}(y)dy \quad x > 0 \end{aligned} \tag{6.9}$$

Using partial integration for $\tau_m \to \infty$:

$$\begin{aligned} V_\infty(x) &= V_\infty(0) + \lambda V_\infty(0) \int_0^x (L_I(x) - L_B(x))dx \\ &+\lambda \int_0^x V_\infty(x-y)(1 - L_B(y))dy \end{aligned} \tag{6.10}$$

Using the normalization equation $V_{\tau_m}(\tau_m) + \int_{\tau_m}^\infty v_{\tau_m}(x)dx = 1$ and the equalities $\int_0^\infty (1 - L_I(x))dx = 1/\mu_I$ and $\int_0^\infty (1 - L_B(x))dx = 1/\mu_B$, it can be written for $x \leq \tau_m$:

$$\begin{aligned} 1 &= cV_\infty(\tau_m) + c\frac{\lambda}{\mu_B}V_\infty(\tau_m) + c\lambda V_\infty(0)\int_{\tau_m}^\infty (L_B(x) - L_I(x))dx \\ &-c\lambda \int_0^{\tau_m} (1 - L_B(x))V_\infty(\tau_m - x)dx \end{aligned} \tag{6.11}$$

Substituting Eq. 6.11 in Eq. 6.10 c can be computed as follows:

$$c = \frac{1}{V_\infty(0) + \frac{\lambda}{\mu_B}V_\infty(\tau_m) + \lambda V_\infty(0)(\frac{1}{\mu_I} - \frac{1}{\mu_B})} \tag{6.12}$$

Substituting the constant c in Eq. 6.8:

$$V_{\tau_m}(x) = \frac{V_\infty(x)}{V_\infty(0) + \frac{\lambda}{\mu_B}V_\infty(\tau_m) + \lambda V_\infty(0)(\frac{1}{\mu_I} - \frac{1}{\mu_B})} \quad 0 \leq x \leq \tau_m \tag{6.13}$$

As mentioned before, $V_\infty(x)$ is the limiting distribution function of the amount of work in the modified system for exceptional service and can be computed using the probability generating function $g(z)$ of the distribution (Π) of the number of packets left behind by the packet under consideration Kleinrock et al. [65], Cooper et al. [66]. The probability p_{Ij} (p_{Bj}) that j packets are arriving at the terminal which is initially in idle (busy) mode during the service is computed from the Laplace-Stieljes transform of service time distributions:

$$p_{Ij} = \int_{\tau=0}^{\infty} \frac{1}{j!} \lim_{z\to 0} \frac{\delta^j}{\delta z^j} L_I((1-z)\lambda, \tau)\, b(\tau)\, d\tau \tag{6.14}$$

$$p_{Bj} = \int_{\tau=0}^{\infty} \frac{1}{j!} \lim_{z\to 0} \frac{\delta^j}{\delta z^j} L_B((1-z)\lambda, \tau) b(\tau)\, d\tau \tag{6.15}$$

where Laplace-Stieljes transform of $d_I(t,\tau)$ and $d_B(t,\tau)$ are computed as:

$$\begin{aligned} L_I(s,\tau) &= \sum_{m=0}^{T} (1-Q_s(\tau))^m Q_s(\tau) exp(-(m+1)\tau s)(\frac{p}{s+p})^m \\ &\quad + (1-Q_s(\tau))^{T+1} exp(-(T+1)\tau s)(\frac{p}{s+p})^T \end{aligned} \tag{6.16}$$

$$\begin{aligned} L_B(s,\tau) &= \sum_{m=1}^{T+1} (1-Q_s(\tau))^{m-1} Q_s(\tau) exp(-m\tau s)(\frac{p}{s+p})^m \\ &\quad + (1-Q_s(\tau))^{T+1} exp(-(T+1)\tau s)(\frac{p}{s+p})^{T+1} \end{aligned} \tag{6.17}$$

The corresponding average service times $1/\mu_I$ and $1/\mu_B$ can be computed by averaging $d_I(t,\tau)$ and $d_B(t,\tau)$ or by differentiating their Laplace-Stieljes transforms:

$$\begin{aligned}
\mu_{Is} &= \int_0^\infty pQ_s(\tau)(1-Q_s(\tau)+p\tau+(1-Q_s(\tau))^T(Q_s(\tau)-1) \\
&\quad (1+TQ_s(\tau)+p\tau(1+Q_s(\tau)+TQ_s(\tau))))^{-1}b(\tau)\,d\tau \qquad (6.18)\\
\mu_I &= \mu_{Is}+\int_0^\infty (1-Q_s(\tau))^{T+1}((T+1)\tau+T/p)\;b(\tau)\,d\tau \qquad (6.19)\\
\mu_{Bs} &= \int_0^\infty \frac{pQ_s(\tau)}{1+p\tau-(1-Q_s(\tau))^T(1+TQ_s(\tau))(1+p\tau)}\,b(\tau)\,d\tau \qquad (6.20)\\
\mu_B &= \mu_{Bs}+\int_0^\infty (1-Q_s(\tau))^{T+1}(T+1)(\tau+1/p)\;b(\tau)\,d\tau \qquad (6.21)
\end{aligned}$$

where $1/\mu_{Is}$ and $1/\mu_{Bs}$ are the average service times of successfully transmitted packets that are generated by idle and busy terminals, respectively.

Let q_{Ij} and q_{Bj} denote the limiting probability distributions of having j packets left behind by the successful packet generated at idle and busy terminals, respectively. The probability of having j packets at a certain user station immediately after completion of a packet transmission ($\Pi_j = q_{Ij} + q_{Bj}$) is derived as the asymptotical equation:

$$\Pi_j = p_{Ij}\Pi_0 + \sum_{m=1}^{j+1} p_{Bj-m+1}\Pi_m \quad j = 0, 1, .. \qquad (6.22)$$

If the probability generating functions $g(z)$, $h_I(z)$ and $h_B(z)$ are defined for probabilities Π, p_I and p_B respectively, the following equation can be written:

$$g(z) = \frac{\Pi_0(zh_I(z) - h_B(z))}{z - h_B(z)} \qquad (6.23)$$

Note that $h_I(z) = L_I(\lambda - \lambda z)$ and $h_B(z) = L_B(\lambda - \lambda z)$. Using the normalization equation $g(1) = 1$ the probability Π_0 that an arbitrary departing packet leaves behind an empty terminal can be found. Since "Poisson arrivals see time averages" [73], this is also the probability (P_0) of finding an empty terminal.

$$\Pi_0 = P_0 = \frac{1 - \lambda/\mu_B}{1 + \lambda(1/\mu_I - 1/\mu_B)} \tag{6.24}$$

Since $g(z) = \Phi(\lambda - \lambda z)$ where Φ is the Laplace-Stieljes transform of the distribution function of the total time (sojourn time) the tagged packet spends in the system, the following equality can be written:

$$g((\lambda - s)/\lambda) \quad = \quad \Phi(s) = (1 - P_0)W_c(s)L_B(s) + P_0 L_I(s) \tag{6.25}$$

where $W_c(s)$ is the Laplace-Stieljes transform of the distribution function of the waiting time given that the system is not empty. Substituting $g(z)$ in Eq. 6.25, Laplace-Stieljes transform ($W(s)$) of the distribution function of the waiting time can be computed:

$$W(s) \quad = \quad \frac{P_0\lambda(1 - L_I(s))}{s - \lambda + \lambda L_B(s)} + P_0 \tag{6.26}$$

The waiting time distribution also represents the limiting distribution $V_\infty(x)$ which is the amount of work in the system. $V_\infty(x)$ is found by taking the inverse Laplace Transform of $W(s)/s$. Note that, $V_\infty(x)/P_0$ corresponds to V_x in Takacs [74] where the author derives the waiting time distribution for served packets only. The parameters of interest π_w, π_r, w_q, w_s and a_r given in Eq. 6.3-6.7 are then computed by substituting $V_\infty(x)$ in Eq. 6.8.

The long run fraction ($P0_{T_m}$) of packets that find the terminal in idle mode and the long run fraction ($P0s_{T_m}$) of only served packets that find the terminal in idle mode can be computed using the limiting distributions for the amount of work in the system:

$$\begin{aligned} P0_{\tau_m} &= V_{\tau_m}(0) && (6.27) \\ P0s_{\tau_m} &= \frac{P_0}{V_\infty(\tau_m)} && (6.28) \end{aligned}$$

Using these probabilities, the long-run average service time (s) of packets and the long-run average delay (w_d) of packets, that are successfully transmitted, can be computed as:

$$\begin{aligned} s &= \frac{P0s_{\tau_m}}{\mu_I} + \frac{1 - P0s_{\tau_m}}{\mu_B} && (6.29) \\ w_d &= w_s(1 - \pi_r) + \frac{P0s_{\tau_m}}{\mu_{Is}} + \frac{1 - P0s_{\tau_m}}{\mu_{Bs}} && (6.30) \end{aligned}$$

6.2.2 Channel Part Analysis

For a finite population of K users, the channel part is modeled as $M/G/\infty//K$ queue model studied in Section 5.2.2 for channel input:

$$G'_{sys} = K\lambda a_r(1 - \pi_w) \qquad (6.31)$$

6.2.3 Combination of the Parts

The user and channel parts are combined as explained in Section 5.2.3 for the following equilibrium equation:

$$K\lambda(1 - \pi_w)(1 - \pi_r) = S' \qquad (6.32)$$

6.3 Numerical Study And Evaluation

This section presents the numerical results obtained from the system analysis, and the simulation results obtained by using Opnet Modeler. As in Chapter 5, the aim of the study performed here is two-fold. First to validate the assumptions used in the analysis and second to study and quantify the effect of packet length distribution on the system performance and demonstrate the use of the analysis as a tool to evaluate and quantify the effect of the various design parameters.

6.3.1 Simulation Model

Before proceeding with the presentation of the results the simulation structure used throughout this section in order to obtain the simulation numerical results is described. The model consists of one base station that serves K mobile users distributed randomly and uniformly in a hexagonal cell (Figure 5.2). Each mobile accepted to the system is assigned a transmission code. The transmission rate is fixed and predetermined. The transmit power is adapted by a control message from the base station whenever there is a change in the number of admitted users. The user sends its packet immediately if its buffer is empty, otherwise it inserts its packet at the tail of the queue. The packet in transmission is removed from the queue when the corresponding positive acknowledgment message from the base station is received or when the maximum number of retransmissions is reached. The packet that did not obtain service for a predefined time is assumed to be lost. The base station computes the interference from other users' packets and the background noise to obtain the SNR for each segment of the packet. The simulations were run for a sufficiently long period of time to collect statistics after the system reaches its steady state.

6.3.2 Numerical Results And Discussion

This section presents the numerical results based on the analysis provided in Section 6.2 and the simulation model described in the previous subsection. For the experiments, the following simulation parameter set is used: $K = 100$ (number of users), $\lambda = 0.05$ (exponential packet arrival rate per user), $A = 2400bits$ (mean arrival packet length), $E_b/N_0 = 10dB$, $G = 60$ (processing gain), $\gamma = 4$ (path loss factor), $R = 4800bits/sec$ (data rate), $t_f = 30MHz$ (base frequency), $W = 288kHz$ (bandwidth). The retransmission rate p is chosen so that the system is stable for the chosen parameter sets. Fixed, exponential and Pareto distributions with cutoff (maximum packet length) are generated.

In order to validate the first assumption that is used throughout the analysis, that the channel load is constant during one bit duration and the number of interfering packets can change at most by 1 between consecutive bits for systems with high transmission rate even if the arrival process is not Poisson, The analytical values obtained based on the expression provided in section 2 are compared with the corresponding simulation results. The results displayed are obtained for Pareto packet length distribution ($k = 990, \alpha = 1.5, cutoff = 12000$) with $T = 10$ and $\tau_m = 5sec$. The probability of having m_{i+1} interferers at the $(i+1)^{th}$ bit when there is m_i interferes at the $(i)^{th}$ bit ($p(m_{i+1}|m_i)$) is displayed in Figure 5.3. The horizontal axis denotes the number of interferers at the i^{th} bit, while the vertical axis denotes the conditional probability of having one less interferer, one more interferer and the same number of interferers at $(i+1)^{th}$ bit from top to bottom respectively. The analytical results are obtained from the Markovian process used in the computation of packet success probability while the simulation results are obtained by counting the number of interfering packets during the transmission of each bit of the tagged packets.

As can be seen from Figure 6.2 simulation and analytical results agree and therefore demonstrate that the number of interfering packets change at most by one between the consecutive bits. The probability $p(m_1)$ of having m_1 interferers for the first bit is displayed in Figure 6.3. The simulation results are computed by counting other users' packets that are transmitted simultaneously during the transmission of the first bit of the tagged packet while analytical values are computed by using (Markovian Model) Poisson distribution with mean $\lambda L/R$.

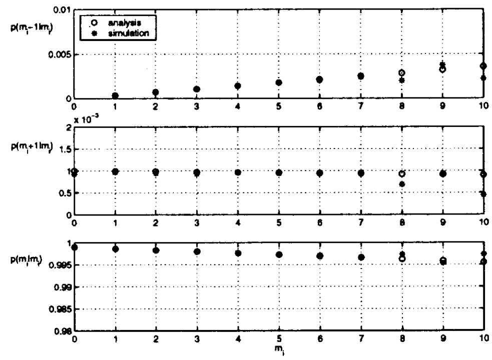


Figure 6.2 Conditional probabilities of number of interferers per bit.

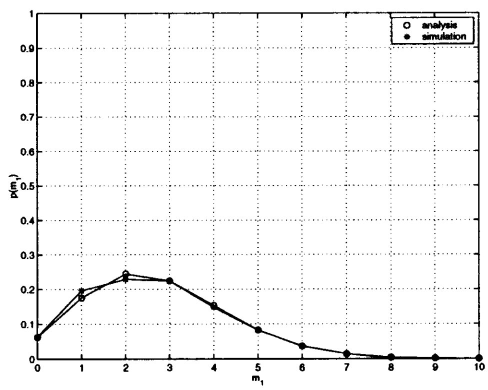


Figure 6.3 Probability of number of interferers for the first bit of the packets.

In order to validate the second assumption that the composite and successful packet duration have the same distribution as that of arrival packet duration (section

2.2) the analytical and simulation results are compared for exponential and Pareto packet length distributions. The graph on the top-left of Figure 6.4 displays the simulation results for the distribution of composite and successful packet size when the arrival packet length distribution is exponential. It can be seen that the composite and successful packet length distributions conform to exponential distributions with different mean values displayed on the top-right of the figure. Since the probability of success is less for packets of longer duration, the composite mean packet length is higher than that of arrival packet length for variable packet length distributions. The same observations hold true when arrival packet length distribution is Pareto with cutoff (bottom of the Figure 6.4). Since the minimum (k) and maximum (cutoff) values of composite and successful packet lengths are the same with those of arrival packet length, the degree (α) of tail heaviness changes between arrival, successful and composite packet length distributions; it is heavier for composite packet length depending on the system load and traffic characteristics. The simulation results displayed on the bottom-right of the figure conform to Pareto distributions with different α values.

In the following, the effect of packet length distribution on the system model is quantified. Specifically, Tables 6.1, 6.2, 6.3, 6.4, 6.5 present the average waiting time (w_q) of lost and served packets, average service time (s) of served packets, average delay (w_d) of successful packets, average number (a_r) of transmissions, packet loss (π_w) due to the restricted waiting time, packet loss (π_r) due to the restricted number of retransmissions, the normalized throughput per second (S') and the normalized input per second (G'_{sys}) to the channel for fixed, exponential and Pareto packet length distributions respectively. As can be seen from the results displayed in Tables 6.1, 6.2, 6.3, 6.4, 6.5 the theoretical and simulation results are very close. The retransmission delay must be longer to provide the stability for the Pareto packet length distribution

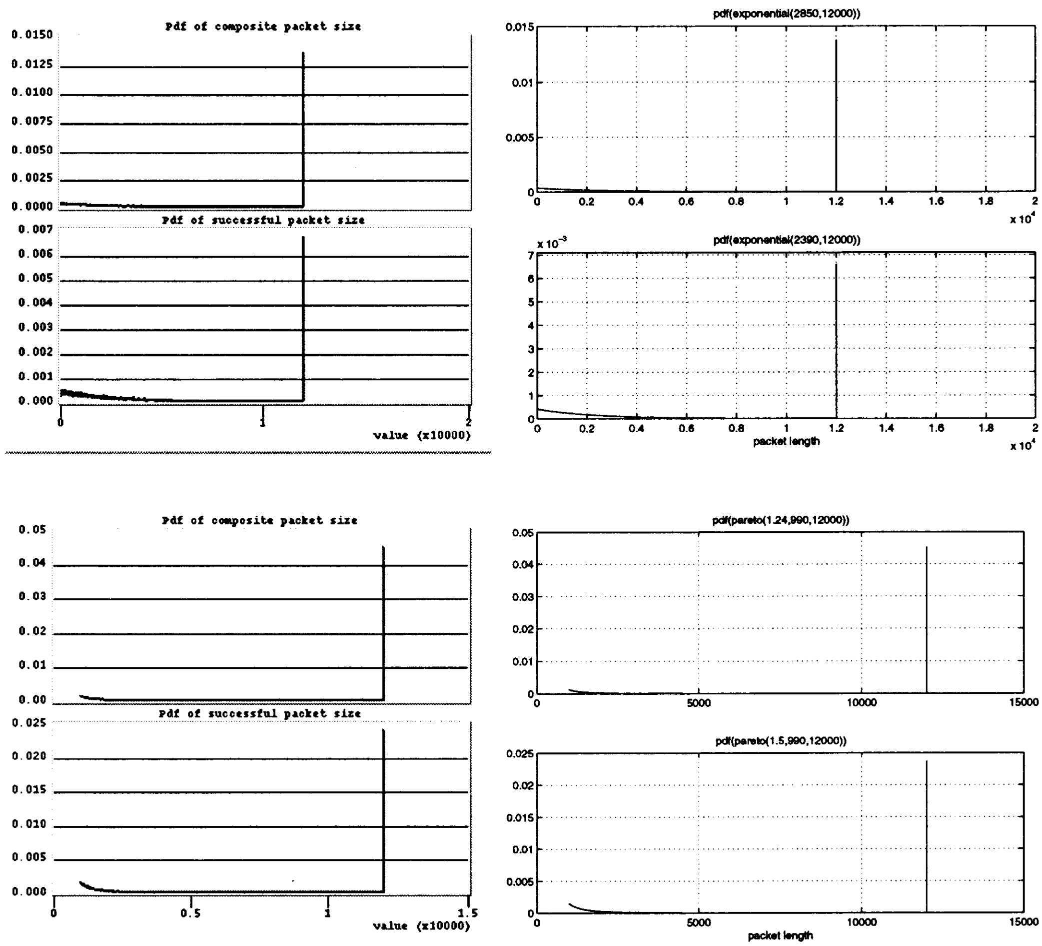


Figure 6.4 Top Left: Simulation results for composite and successful packet length distribution for Exponential arrival packet length ($B_c = 2760$); Top Right: Exponential distributions with mean 2760 and 2400; Bottom Left: Simulation results for composite and successful packet length distribution for Pareto arrival packet length ($Pareto(k = 990, \alpha = 1.24, cutoff = 12000), Bc = 2848$ with $Pareto(k = 990, \alpha = 1.5, cutoff = 12000)$); Bottom Right: Pareto distributions with parameters $k = 990, cutoff = 12000, \alpha = 1.24$ and $\alpha = 1.5$.

	$T=1, \tau_m = 5$ sec		$T=10, \tau_m = 5$ sec	
	Analysis	Simulation	Analysis	Simulation
w_q	0.97	0.93	1.08	1.04
s	5.38	5.436	6.02	5.92
w_d	4.98	5.01	6.15	6.04
π_w	0.172	0.17	0.1914	0.19
π_r	0.0188	0.0185	0	0
a_r	1.137	1.136	1.158	1.153
G'_{sys}	4.71	4.69	4.68	4.65
S'	4.062	4.06	4.043	4.04

Table 6.1 Results for fixed packet length ($p = 1/25$)

	$T=1, \tau_m = 5$ sec		$T=10, \tau_m = 5$ sec	
	Analysis	Simulation	Analysis	Simulation
w_q	1.03	0.978	1.35	1.322
s	5.63	5.75	7.84	7.76
w_d	4.52	4.76	7.99	7.896
π_w	0.179	0.18	0.2475	0.246
π_r	0.031	0.036	0	0
a_r	1.143	1.145	1.219	1.22
G'_{sys}	4.69	4.7	4.59	4.60
S'	3.97	3.96	3.76	3.77

Table 6.2 Results for exponential packet length ($p = 1/25, cutoff = 12000$)

with heavier tail due to the higher probability of having longer packets (Tables 6.4 and 6.5). When the distribution has light tail and cutoff parameter is low, the difference between the results (Tables 6.1, 6.2, 6.3 and 6.4) is small. However, when the cutoff parameter is large (Table 6.5), a severe degradation of the system performance is observed.

As mentioned before, the analysis can be used in order to provide an in depth evaluation of the system behavior under various design parameters. For instance, Figure 6.5 displays the performance results when different data rates and

	$T = 1, \tau_m = 5$ sec		$T = 10, \tau_m = 5$ sec	
	Analysis	Simulation	Analysis	Simulation
w_q	1.04	0.981	1.419	1.37
s	5.65	5.85	8.34	8.12
w_d	4.61	4.89	8.49	8.287
π_w	0.18	0.181	0.26	0.256
π_r	0.031	0.0359	0	0
a_r	1.142	1.145	1.235	1.233
G'_{sys}	4.68	4.688	4.57	4.58
S'	3.97	3.95	3.7	3.72

Table 6.3 Results for exponential packet length ($p = 1/25, cutoff = 24000$)

	$T = 1, \tau_m = 5$ sec		$T = 10, \tau_m = 5$ sec	
	Analysis	Simulation	Analysis	Simulation
w_q	0.987	0.95	1.33	1.3
s	5.39	5.6	7.56	7.68
w_d	4.37	4.83	7.92	7.89
π_w	0.179	0.175	0.2433	0.242
π_r	0.029	0.03	0	0
a_r	1.148	1.141	1.238	1.216
G'_{sys}	4.712	4.71	4.615	4.61
S'	3.98	4.0	3.78	3.79

Table 6.4 Results for Pareto packet length ($p = 1/25, Pareto(k = 990, \alpha = 1.5, cutoff = 12000)$)

	$T = 1, \tau_m = 5$ sec		$T = 10, \tau_m = 5$ sec	
	Analysis	Simulation	Analysis	Simulation
w_q	1.32	1.28	1.99	1.91
s	7.61	7.86	13.52	12.83
w_d	5.96	6.64	13.68	13.01
π_w	0.249	0.243	0.379	0.367
π_r	0.0275	0.027	0	0
a_r	1.129	1.126	1.241	1.238
G'_{sys}	4.24	4.26	3.91	3.918
S'	3.65	3.68	3.15	3.165

Table 6.5 Results for Pareto packet length ($p = 1/40, Pareto(k = 921, \alpha = 1.5, cutoff = 24000)$)

processing gains are applied to the system with Pareto packet length distribution with parameters $k = 990, \alpha = 1.5, cutoff = 12000$ and $T = 10, \tau_m = 5sec$. The data rate is changed while other system parameters are kept constant. Increasing the transmission rate of common channel, one one hand decreases transmission time of a packet but on the other hand it increases the noise level, therefore increasing the number of retransmissions. Figure 6.5 displays the results for packet loss, normalized throughput, average delay of the successfully transmitted packets and average waiting time of lost packets, that are computed in order to find optimal channel rate that minimizes packet loss and delay while increasing the system throughput and utilization.

In this section, an analytical method and model to study and evaluate the performance of CDMA-CPCH schemes is provided. Specifically, a system model with finite population, restricted sojourn time and variable packet length is considered in order to evaluate a realistic wireless data network. Based on the analysis an in depth study of the performance related parameters of the system is presented. The emphasis of this study is placed on the investigation of the effect of the packet length distribution to the system characteristics. It is shown via analysis and simulation that for the packet length distributions with the same mean but different tail properties the system behavior can change dramatically. Since various packet length distributions, such as the ones with heavy tailed packet can cause bottlenecks to the system, it is critical to implement dynamic control.

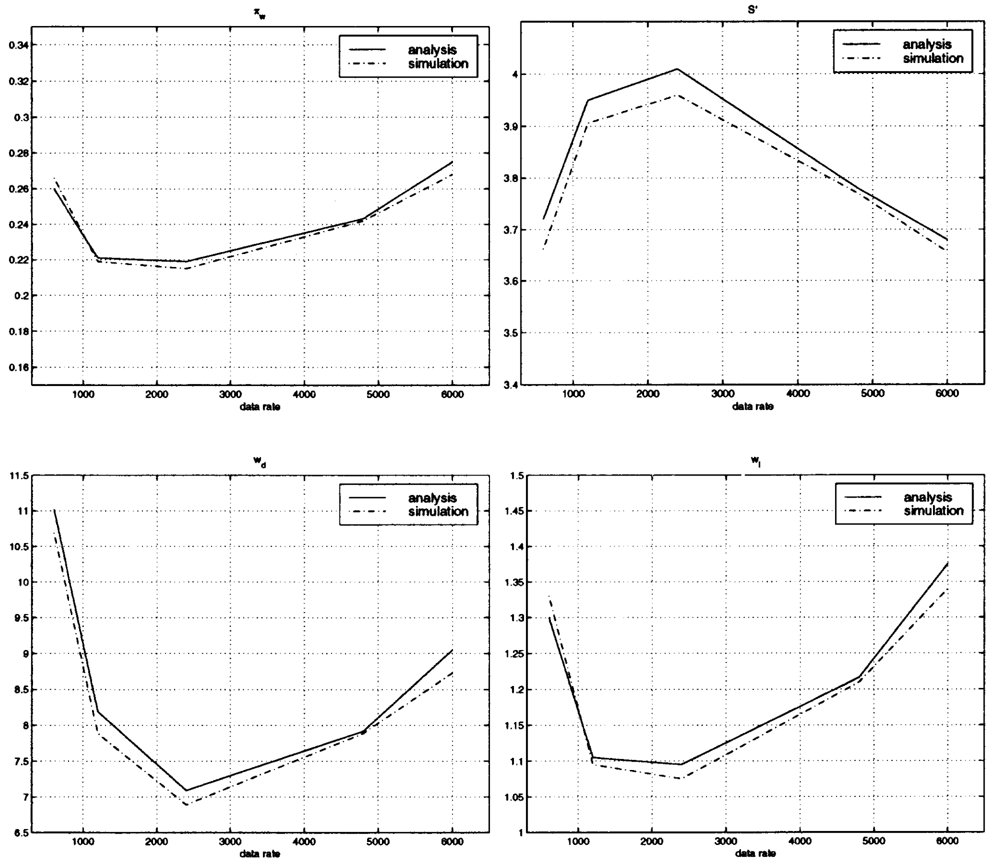


Figure 6.5 Top Left: Packet loss; Top Right: Normalized throughput; Bottom Left: Average delay of the successfully transmitted packets; Bottom Right: Average waiting time of lost packets for different data rates.

The analysis presented in this section provides a first and critical step towards the direction of implementation of an adaptive and dynamic system control mechanism. Furthermore, the study provides an analytical tool to find optimum variable system parameters (e.g., channel rate, maximum packet length to be transmitted, number of users admitted to system) when the fixed system parameters (e.g., bandwidth) and QoS requirements of data (e.g., delay and loss bounds) are given. Since the analysis is done with respect to both completely served and lost packets it also quantifies the system utilization. It enables to quantify the total system utilization such as the buffer time occupied by the packets lost due to the impatience factor and the service time of packets lost due to the limit on the number of retransmissions.

CHAPTER 7

ANALYSIS OF CDMA SYSTEMS WITH MULTIMEDIA SERVICES

In this chapter, in order to support multimedia services using common infrastructure dedicated mode transmission for real-time services and common channel transmission for non real-time services with varying processing gain studied in Chapters 5 and 6 are considered. Specifically, two types of services are investigated: real-time voice and nonreal-time data services with different traffic characteristics and quality of service requirements. The corresponding system model is represented in Figure 7.1. Each user is assigned a unique transmission code. A 20 ms framing structure is used for voice packets for a frame error rate less than 0.01 without retransmissions, therefore they are transmitted in dedicated circuit mode. A variable frame size for data users is used with automatic repeat request (ARQ). When data packets are received with errors due to the multiuser interference and background noise, they are retransmitted after a random delay. Hence, the aggregated channel input consists of new voice and data packets and retransmitted data packets.

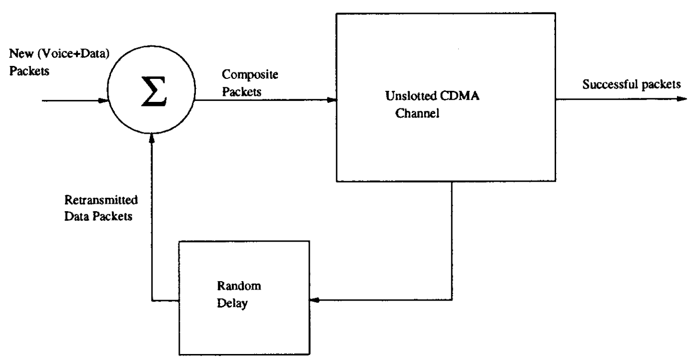


Figure 7.1 Schematic Representation of CDMA Channel

7.1 System Model

Figure 4.16 represents an unslotted spread spectrum packet radio network with a finite number of voice and data users. The traffic model for voice users is exponentially distributed on-off model as given in Section 3.

A new user accesses the system by sending an access message through the access channels. The number of access channels is assumed to be high so that access failure is omitted. Then, the user transmits its packets through the assigned channel (code). For data users, packet interarrival time is assumed to be exponentially distributed with mean $1/\lambda$. The analysis is done for a general packet length distribution. The numerical results presented in this chapter are obtained for fixed, Exponential and Pareto distributed packet lengths with cutoff (maximum packet length) (Section 3).

7.2 System Analysis

7.2.1 Performance Analysis for Data Users

Assuming that the system condition changes slowly enough to compute the steady state probabilities, the system is divided into user and channel part. Since the user part analysis is the same as in the Chapters 5 and 6, only the channel part analysis is given.

7.2.1.1 Channel Part Analysis For a finite population of K data users, the channel part can be modeled as $M/G/\infty//K$ queue model which is invariant to the service time distribution Cooper et al. [66].

The average number of data packets G_{sys} transmitted to the channel during a data packet duration and normalized average number of data packets G'_{sys} transmitted in 1 second are given by:

$$G_{sys} = \frac{K\lambda_c/\mu_c}{1+\lambda_c/\mu_c} \tag{7.1}$$

$$G'_{sys} = G_{sys}\,\mu_c \tag{7.2}$$

where $\mu_c = R/B_c$ and λ_c is actual birth rate.

The number of packets transmitted to the channel per unit time is equal to the number of total transmissions of packets can be computed from Eq. 5.25 and 6.31 for cases of finite buffer and finite sojourn time respectively.

The birth rate (λ_c) for actually transmitted packets can then be computed from Eq. 7.1-7.2.

It is assumed that the channel load (voice and data) is constant during one data bit duration (Δt_d) of a data packet and that the number of interfering data and voice packets can change at most by 1 between the consecutive bits.

The probability of having v voice interferers at the the first bit of a data packet is:

$$p_v^d(j) = \binom{V}{j}\xi^j(1-\xi)^{V-j} \tag{7.3}$$

The probability of having d data interferers at the the first bit of a data packet is:

$$p_d^d(j) = \binom{K-1}{j}\frac{(\lambda_c/\mu_c)^j}{(1+\lambda_c/\mu_c)^{K-1}} \tag{7.4}$$

The conditional probabilities of the number of voice and data interferers at consecutive bits of a data packet is given as:

$$p^d(v_i|v_i) = 1 - v\mu_v\Delta t_d - (V - v)\lambda_v\Delta t_d \quad (7.5)$$

$$p^d(v_i|v_i + 1) = (v + 1)\mu_v\Delta t_d \quad (7.6)$$

$$p^d(v_i|v_i - 1) = (V - v + 1)\lambda_v\Delta t_d \quad (7.7)$$

$$p^d(d_i|d_i) = 1 - d\mu_c\Delta t_d - (K - d - 1)\lambda_c\Delta t_d \quad (7.8)$$

$$p^d(d_i|d_i + 1) = (d + 1)\mu_c\Delta t_d \quad (7.9)$$

$$p^d(d_i|d_i - 1) = (K - d)\lambda_c\Delta t_d \quad (7.10)$$

The probability that the packet is transmitted successfully from the first bit to the $(i-1)^{th}$ bit with $(d+v)$ interfering packets on the i^{th} bit is $P_s(d, v, i)$:

For $i = 1 : P_s(d, v, 1) = p_v^d(v)p_d^d(d)$.

For $i > 1$:

$$\begin{aligned}
P_s(d,v,i) = {} & P_s(d,v,i-1)p^d(v_i|v_i)p^d(d_i|d_i)(1 - P_b(d,v)) + \\
& P_s(d+1,v,i-1)p^d(v_i|v_i)p^d(d_i|d_i+1)(1 - P_b(d+1,v)) + \\
& P_s(d-1,v,i-1)p^d(v_i|v_i)p^d(d_i|d_i-1)(1 - P_b(d-1,v)) + \\
& P_s(d,v+1,i-1)p^d(v_i|v_i+1)p^d(d_i|d_i)(1 - P_b(d,v+1)) + \\
& P_s(d,v-1,i-1)p^d(v_i|v_i-1)p^d(d_i|d_i)(1 - P_b(d,v-1)) + \\
& P_s(d+1,v+1,i-1)p^d(v_i|v_i+1)p^d(d_i|d_i+1)(1 - P_b(d+1,v+1)) + \\
& P_s(d-1,v-1,i-1)p^d(v_i|v_i-1)p^d(d_i|d_i-1)(1 - P_b(d-1,v-1)) + \\
& P_s(d+1,v-1,i-1)p^d(v_i|v_i-1)p^d(d_i|d_i+1)(1 - P_b(d+1,v-1)) + \\
& P_s(d-1,v+1,i-1)p^d(v_i|v_i+1)p^d(d_i|d_i-1)(1 - P_b(d-1,v+1)) \quad (7.11)
\end{aligned}$$

The bit error rate ($P_b(d, v)$) for bpsk modulation is expressed as:

$$P_b(d,v) = Q(\sqrt{SNR_d}) \tag{7.12}$$

where $Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^{\infty} exp(-u^2/2)du$ and SNR_d is the signal to noise ratio for data users and computed as:

$$SNR_d(dB) = 10log(G_d) + 10log(p_d/(dp_d + v + No/P)) \tag{7.13}$$

where G_d is the number of chips per bit, d is the number of interfering data packets, v is the number of interfering voice packets, p_d is the ratio of data received power to the voice received power (P), E_b is the bit energy of the signal and No is spectral density of AWGN.

The throughput S in a packet duration is computed using packet success probability Qs:

$$S = \int_{\tau=0}^{\infty} G_{sys} Qs(\tau)\; b_c(\tau)\, d\tau \tag{7.14}$$

$$Qs(\tau) = \sum_{v=0}^{V} \sum_{d=0}^{K-1} P_s(d,v,L(\tau))(1 - P_b(d,v))Pv(v) \tag{7.15}$$

where $L(\tau) = \tau R_d$. The mean number of successful packets in 1 second is then computed as $S' = S\,\mu_c$.

A search loop is formed for λ_c and mean composite packet length (B_c) to compute the throughput using channel part model given in chapters 5 and 6.

7.2.2 Performance Analysis for Voice Users

As in the previous section, it is assumed that the channel load (voice and data) is constant during one bit duration (Δt_v) of a voice packet and that the number of interfering data and voice packets can change at most by 1 between the consecutive bits. This assumption is shown to be valid via simulation in the next section.

The probability of having v voice interferers at the the first bit of a voice packet is:

$$p_v^v(j) = \binom{V-1}{j} \xi^j (1-\xi)^{V-j-1} \tag{7.16}$$

The probability of having d data interferers at the the first bit of a voice packet is:

$$p_d^v(j) = \binom{K}{j} \frac{(\lambda_c/\mu_c)^j}{(1+\lambda_c/\mu_c)^K} \tag{7.17}$$

Note that the birth rate (λ_c) and death rate (μ_c) of data packets is the rate values for composite (new+retransmitted) packets.

The conditional probabilities of the number of voice and data interferers at consecutive bits of a voice packet is given as:

$$p^v(v_i|v_i) = 1 - v\mu_v\Delta t_v - (V-v-1)\lambda_v\Delta t_v \tag{7.18}$$

$$p^v(v_i|v_i+1) = (v+1)\mu_v\Delta t_v \tag{7.19}$$

$$p^v(v_i|v_i-1) = (V-v)\lambda_v\Delta t_v. \tag{7.20}$$

$$p^v(d_i|d_i) = 1 - d\mu_c\Delta t_v - (K-d)\lambda_c\Delta t_v \tag{7.21}$$

$$p^v(d_i|d_i+1) = (d+1)\mu_c\Delta t_v \tag{7.22}$$

$$p^v(d_i|d_i-1) = (K-d+1)\lambda_c\Delta t_v \tag{7.23}$$

The probability that the voice packet has $(d+v)$ interfering data and voice packets on the i^{th} bit is computed as:

For $i = 1 : p_{dv}(d, v, i) = p_v^v(v)p_d^v(d)$

For $i > 1$:

$$\begin{aligned}
p_{dv}(d,v,i) &= p_{dv}(d,v,i-1)p^v(v_i|v_i)p^v(d_i|d_i) + \\
&\quad p_{dv}(d+1,v,i-1)p^v(v_i|v_i)p^v(d_i|d_i+1) + \\
&\quad p_{dv}(d-1,v,i-1)p^v(v_i|v_i)p^v(d_i|d_i-1) + \\
&\quad p_{dv}(d,v+1,i-1)p^v(v_i|v_i+1)p^v(d_i|d_i) + \\
&\quad p_{dv}(d,v-1,i-1)p^v(v_i|v_i-1)p^v(d_i|d_i) + \\
&\quad p_{dv}(d-1,v-1,i-1)p^v(v_i|v_i-1)p^v(d_i|d_i-1) + \\
&\quad p_{dv}(d+1,v+1,i-1)p^v(v_i|v_i+1)p^v(d_i|d_i+1) + \\
&\quad p_{dv}(d+1,v-1,i-1)p^v(v_i|v_i-1)p^v(d_i|d_i+1) + \\
&\quad p_{dv}(d-1,v+1,i-1)p^v(v_i|v_i+1)p^v(d_i|d_i-1) \qquad (7.24)
\end{aligned}$$

The bit error rate $(P_b(k))$ for bpsk modulation is expressed as:

$$P_b(d,v) = Q(\sqrt{SNR_v}) \qquad (7.25)$$

where $Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty exp(-u^2/2)du$ and SNR_v is the signal to noise ratio for voice users and computed as:

$$SNR_v(dB) = 10log(G_v) + 10log(1/(dp_d + v + No/P)) \qquad (7.26)$$

where G_v is the number of chips per bit, d is the number of interfering data packets, v is the number of interfering voice packets, p_d is the ratio of data received power to the voice received power (P), E_b is the bit energy of the signal and No is spectral density of AWGN.

7.3 Numerical Examples for Finite Population CDMA-CPCH Systems with Finite Data Buffers

This section presents the numerical results from the system analysis and the simulation results obtained by using Opnet Modeler.

7.3.1 Simulation Model

The model consists of one base station that serves $(K + V)$ mobile users distributed randomly and uniformly in a hexagonal cell (Figure 5.2). Each mobile accepted to the system is assigned a transmission code. Each user within the same class has the same data rate and received power. The quality of service for voice users is assumed to be guaranteed during one packet transmission (with a probability $1 - P_{lv}$) while it is not guaranteed for data users. The voice user sends its packets immediately and transmits only one time. The data user sends its packet immediately if its buffer is empty, otherwise it inserts its packet at the tail of the queue if the buffer size is not exceeded. The packet in transmission is removed from the queue when the corresponding positive acknowledgment message from the base station is received. For each received packet, the base station computes the interference from other users' packets and the background noise to obtain the SNR for each segment of the packet. This is done by using the radio link pipelines of OPNET modeler. The simulations were run for a sufficiently long period of time to collect statistics after the system reaches its steady state.

7.3.2 Numerical Results

This section provides the numerical results based on the analysis and the simulation model described in the previous subsection, in order to validate the assumptions used in the analysis and to investigate the impact of various model parameters on the system performance. For these purposes, the following simulation parameter set is used: $K = 50$ (number of data users), $V = 10$ (number of voice users), $\xi = 0.4$ (voice activity factor) with $1/\kappa = 1s$ (voice mean on time) and $1/\beta = 1.5s$ (voice mean off time), $R_v = 9600bps$ (voice channel rate), $L_v = 192bits(20ms)$ (voice packet length), $\Gamma_v = 7dB$ (SNR requirement for voice packets), $A = 500bits$ (mean arrival data packet length), $\gamma = 4$ (path loss factor), $R_d = 32000bps$ (data channel rate), $t_f = 30MHz$ (base frequency), $W = 1.25MHz$ (bandwidth). The retransmission rate p is chosen so that the system is stable for the chosen buffer sizes (B) and data arrival rates (λ).

First the simulation results are used to validate the assumption that the channel load is constant during one bit duration and that the number of interfering data and voice packets can change at most by 1 between consecutive bits of voice packets. Note that the same statements hold for data packets since one data bit duration is comparable to one voice bit duration when compared to birth and death rate of multimedia packets. Figure 7.2 displays the simulation and analytical results (Eq. 7.18-7.23) of the probability of the number of voice and data interferers at the first bit of a voice packet for $\lambda = 1/0.5$, $p = 1/0.6$, $B = 3$. The simulation results are obtained by counting the number of interfering voice and data packets at the first bits of voice packets.

From Figures 7.3 and 7.4 it can be seen that the probability of voice and data interferers during a voice packet transmission can be computed using the probability given by Eq. 7.24. The probability of having d_{i+1} data (v_{i+1} voice) interferers at the $(i+1)^{th}$ bit when there is d_i data (v_i voice) interferes at the $(i)^{th}$ bit is displayed in Figure 7.3 (Figure 7.4). The horizontal axis denotes the number of interferers at the i^{th} bit, while the vertical axis denotes the conditional probability of having one less interferer, one more interferer and the same number of interferers at $(i+1)^{th}$ bit from bottom to top respectively. The analytical results are obtained from the

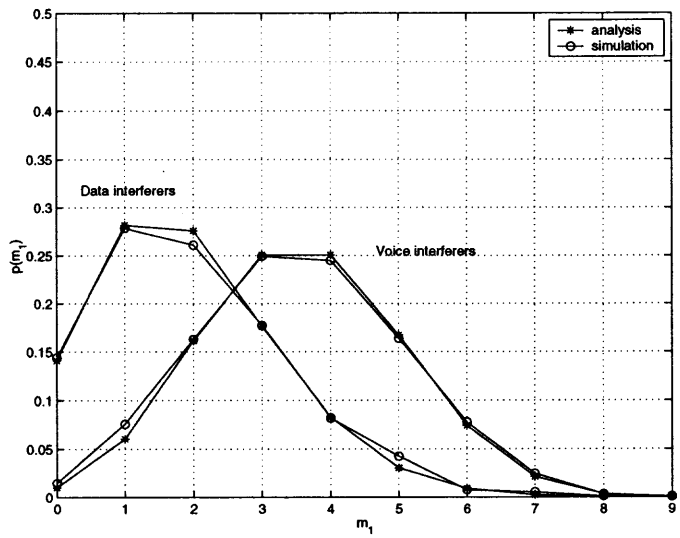


Figure 7.2 Probability of the number of voice and data interferers at the first bit of a voice packet.

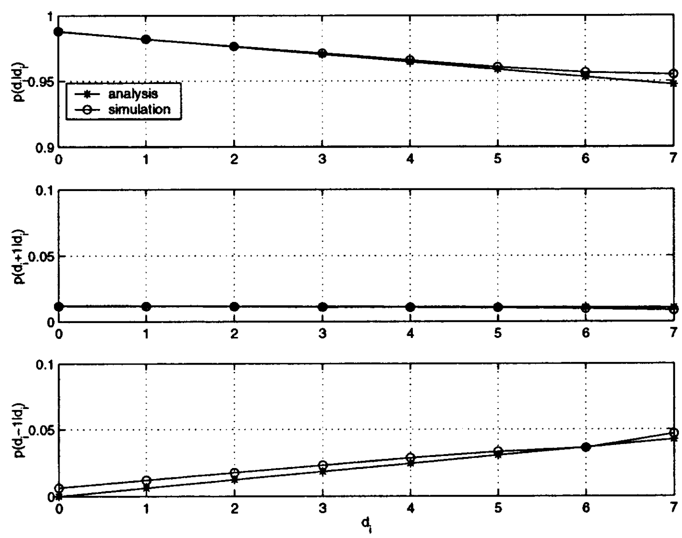


Figure 7.3 Change of the number of data interferers between consecutive bits of a voice packet.

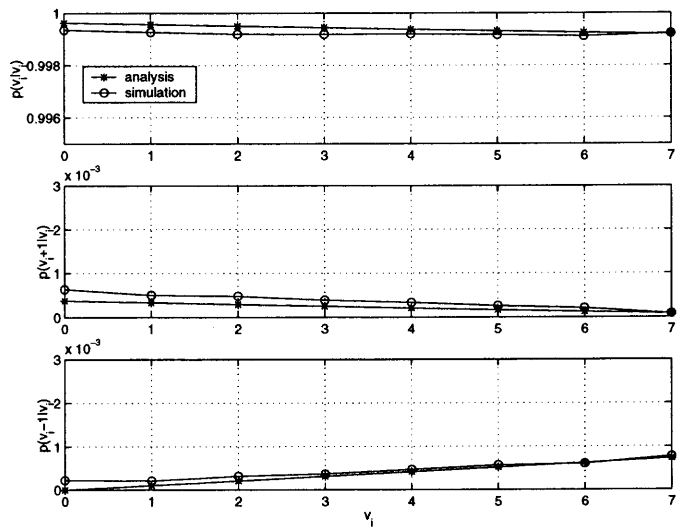


Figure 7.4 Change of the number of voice interferers between consecutive bits of a voice packet.

Markovian process (Eq. 7.24) while the simulation results are obtained by counting the number of interfering packets during the transmission of each bit of the tagged voice packets. As can be seen from the figures simulation and analytical results agree and demonstrate that the number of interfering packets change at most by one between the consecutive bits.

Figure 7.5 displays Opnet results for composite data packet length when the arrival data packet length is distributed according to Pareto distribution with parameters $Pareto(k = 145, \alpha = 1.3, cutoff = 12000)$ in order to show the validation of the assumption that the composite packet duration has the same distribution as that of arrival packet duration. It can be seen that the simulation result for composite packet length conforms to a Pareto distribution with heavier tail properties. Since the minimum (k) and maximum (cutoff) values of composite packet length are the same with those of arrival packet length, the degree (α) of tail heaviness changes between arrival and composite packet length distributions;

it is heavier for composite packet length depending on the system load and traffic characteristics.

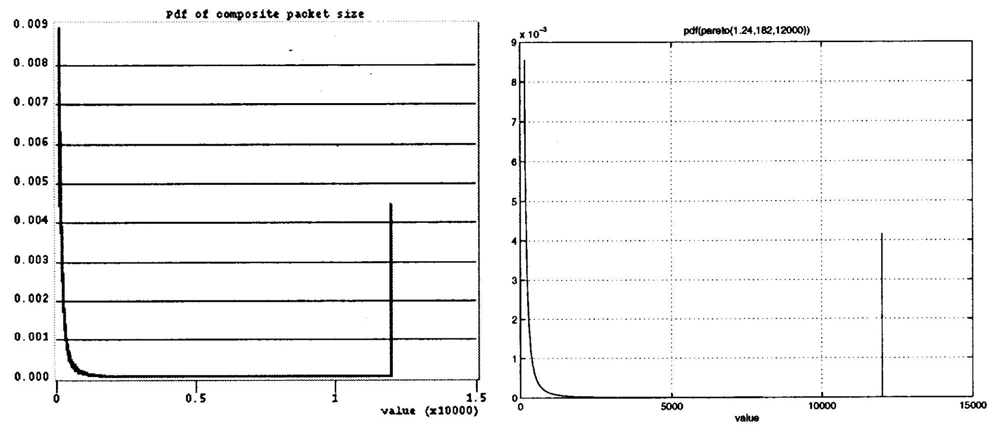


Figure 7.5 Left: Simulation result for composite packet length distribution for Pareto arrival packet length with parameters $Pareto(k = 145, \alpha = 1.3, cutoff = 12000)$; Right: Pareto distribution with parameters $k = 145, \alpha = 1.24, cutoff = 12000$ ($\lambda = 1/0.5$, $p = 1/0.6$, $B = 3$).

In the following, the effect of packet length distribution on the system model is demonstrated. Tables 7.1-7.2 present the performance of data users, specifically the normalized input per second (G'_{sys}) to the channel, the normalized throughput per second (S'), the mean (D) and standard deviation (σ_D) of packet delay, average service time (s) and average waiting time (w) of packets, the probability (P_0) to find data user idle, average queue length (Q) at each user's buffer, packet rejection probability (Q_r) and average transmission number (a_r) per packet for fixed, exponential and Pareto packet length distributions respectively. The examples are chosen so that the probability ($P_{lv} = Prob(SNR_v > \Gamma_v)$) that a voice user exceeds the voice SNR threshold is smaller than 0.01. An example for the computation of SNR_v is given in Figure 7.6. The analytical results (from Eq. 7.24) are obtained by computing the time interval that a voice packet has a certain SNR value while simulation results are obtained from the corresponding histogram. Note that the vertical axis denotes $SNR_v(dB) - 10log(G_v)$.

	Fixed		Exponential		Pareto($k = 145, \alpha = 1.3$)	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
G'_{sys}	94.356	94.23	93.93	94.00	93.83	93.99
S'	93.08	92.94	92.58	92.597	92.24	92.48
D	0.2138	0.2141	0.22	0.222	0.226	0.224
σ_D	0.61	0.609	0.625	0.621	0.644	0.626
s	0.107	0.1069	0.1099	0.11	0.1106	0.1099
w	0.107	0.1072	0.1108	0.112	0.115	0.114
P_0	0.796	0.795	0.796	0.795	0.792	0.796
Q	0.398	0.398	0.408	0.410	0.417	0.414
Q_r	0.071	0.0708	0.0735	0.074	0.076	0.075
a_r	1.0136	1.0138	1.0146	1.01	1.0167	1.016

Table 7.1 Results for $\lambda = 1/0.5$, $p = 1/0.6$, $B = 3$, $p_d = 2$, $cutoff = 12000$.

While at light traffic load, the performance is similar for different packet length distributions (Table 7.1), the degradation is severe for Pareto packet length distribution when the offered load is high (Table 7.2). The stability of the system requires higher retransmission delay for heavy tailed packet length distribution resulting higher packet delay and less throughput. Note that, all other parameters and the mean values of offered traffic is the same for the displayed results, which demonstrates the effect of higher moments of the traffic load on the system performance.

The following set of experiments demonstrates that the analytical tool can be used to develop and analyze optimal resource allocation strategies to maximize throughput of wireless systems with integrated services. Given the multimedia

	Fixed		Exponential		Pareto($k = 145, \alpha = 1.3$)	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
G'_{sys}	188.07	188.12	185.74	185.12	111.21	111.78
S'	173.56	173.39	167.73	167.32	108.318	109.21
D	0.407	0.411	0.445	0.449	0.978	0.971
σ_D	0.56	0.55	0.6	0.58	1.163	1.161
s	0.179	0.18	0.192	0.194	0.375	0.373
w	0.229	0.23	0.253	0.255	0.6	0.598
P_0	0.379	0.375	0.354	0.349	0.186	0.189
Q	1.414	1.426	1.494	1.504	2.127	2.121
Q_r	0.305	0.306	0.329	0.33	0.566	0.565
a_r	1.084	1.085	1.1074	1.1064	1.0266	1.024

Table 7.2 Results for $\lambda = 1/0.2$, $p = 1/0.3$ for fixed and exponential packet length, $p = 1/0.6$ for fixed and Pareto packet length, $B = 3$, $p_d = 2$, $cutoff = 12000$.

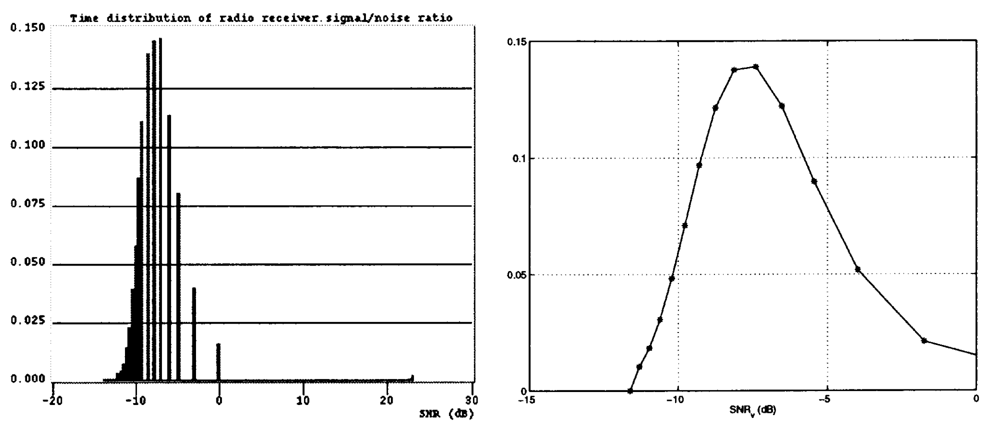


Figure 7.6 Simulation (left) and analytic (right) results of time distribution of voice packets SNR (fixed data packet length, $\lambda = 1/0.2$, $p = 1/0.3$)

user requirements and system parameters, the allocation problem addressed can be formulated as a constrained optimization problem. Let N be the number of users of different service classes in the system. The power and rate allocation vectors are defined for the N users to be $\mathbf{P} = [P_1, P_2, ..., P_N]$ and $\mathbf{R} = [R_1, R_2, ..., R_N]$ respectively. The tolerable QoS for each user is in the forms of maximum bit error rate (BER) or signal to noise ratio (SNR) denoted by $\mathbf{\Gamma} = [\Gamma_1, \Gamma_2, ..., \Gamma_N]$ and maximum delay denoted by $\mathbf{D^{max}} = [D_1^{max}, D_2^{max}, ...D_N^{max}]$. The objective of the admission control and resource allocation policy is to determine the channel allocation to the users with the corresponding feasible power and rate vectors such that overall traffic carried by the system is maximized when all QoS constraints are met with equality.

An experiment for optimal power allocation is presented with analytical and simulation results in order to show the efficiency of the analytical results. More specifically, Figure 7.7 displays performance results in order to find optimal power allocation of integrated services, e.g. optimal ratio (p_d) of received data power to the received voice power. Although this system has no closed form solution for optimal power vector, the power allocation can be optimized by using the analytical results for integrated services for the more general and realistic case with finite population of voice and data users and finite buffers and variable packet length for data users. For instance, the first graph in Figure 7.7 displays the results of mean (D) packet delay for different p_d values. A tremendous improvement is observed when the received power of data users is doubled as the delay decreases by 88%. However, the improvement for data users is very low for $p_d > 3$ while performance degradation of voice users increases.

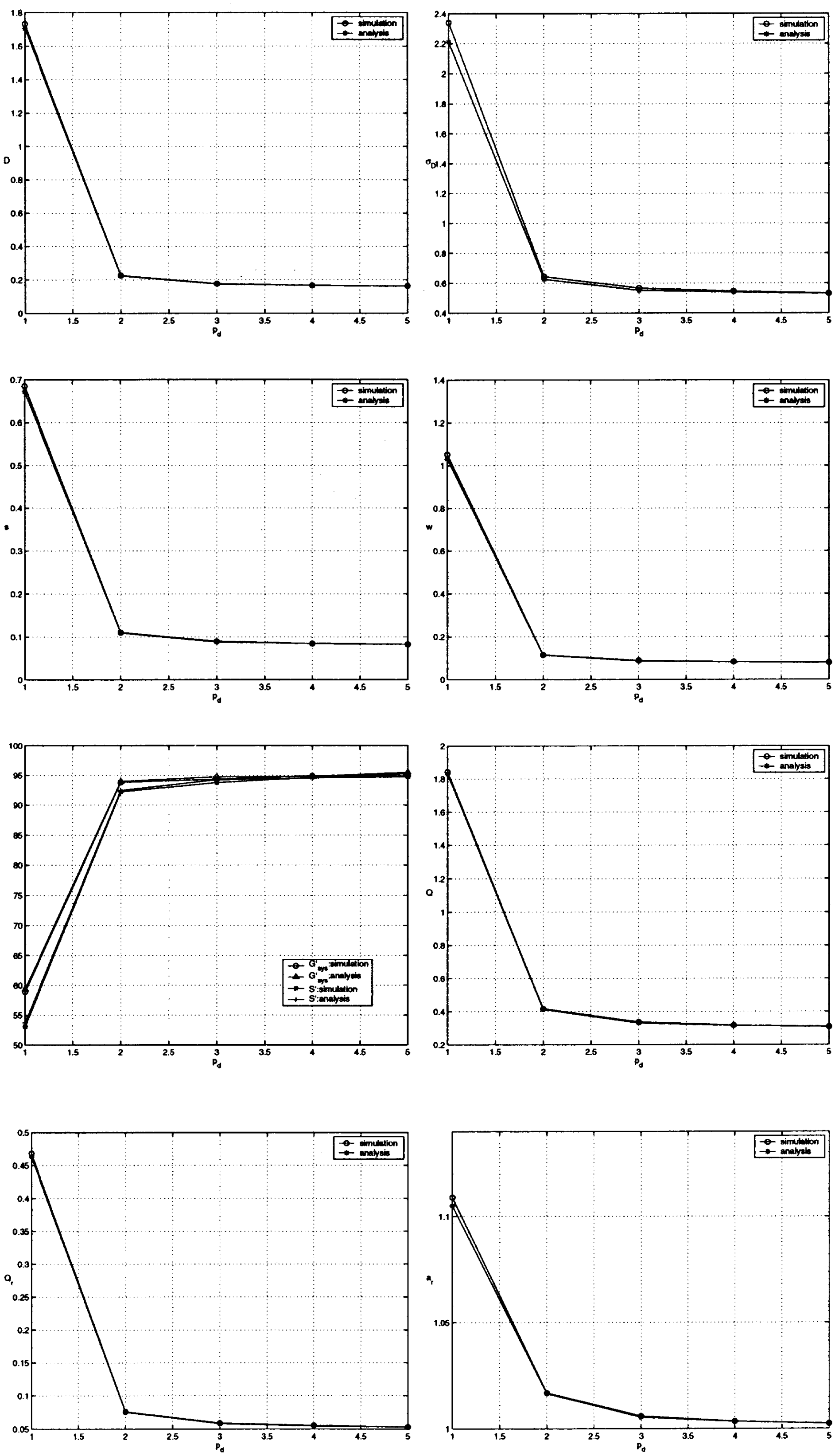


Figure 7.7 Results for Pareto packet length (k = 145, α = 1.3, $cutoff$ = 12000, $\lambda = 1/0.5$, $B = 3$, $p = 1/1.1$ for $p_d = 1$ and $p = 1/0.6$ elsewhere)

CHAPTER 8

CONCLUSIONS

This thesis is a comprehensive study of resource allocation strategies to maximize throughput of wireless systems with multimedia services. The thesis focuses in the problem of optimal transmission schemes in wireless CDMA networks for services with different traffic characteristics and QoS requirements.

The thesis first examines different switching techniques for data applications, namely circuit, dedicated burst/packet transmission modes and common packet channel transmission schemes. Chapter 4 discusses the advantages of burst switching technique for bursty data traffic for unslotted and slotted CDMA networks respectively. The comparison of burst mode transmission and common channel transmission schemes is done for integrated services by addressing the effect of data traffic characteristics on the system performance. It is shown that the advantage of burst reservation schemes for data services is the minimization of the interference for voice and data packets at the expense of higher complexity to control and measure the channel load. Aloha type common packet transmission requires higher rate of retransmission for data users while a simpler control mechanism is needed.

In Section 5, the performance of CDMA common packet channel transmission schemes is studied in depth for finite population and finite buffers for more realistic data traffic characteristics than assumed in the literature. The performance analysis demonstrates that for packet length distributions with the same mean but different tail properties the system behavior can change dramatically. For the same average traffic load, packet delay and packet loss are increased while system throughput is decreased for heavy-tailed packet length distributions where unacceptable packet delays are observed. In Section 6, the system behavior is analyzed by restricting the time a packet can spend in the system, specifically by limiting the waiting and serving time of a packet. Furthermore, it is shown that the study provides an analytical tool to find optimum variable system parameters (e.g., channel rate, maximum packet length to be transmitted, number of users admitted to system) when the fixed system parameters (e.g., bandwidth) and QoS requirements of data (e.g., delay and loss bounds) are given.

Finally, the performance of wireless CDMA systems for integrated services is analyzed by studying the effect of each service on the other. In Chapter 7, in order to support multimedia services using common infrastructure, dedicated mode transmission is applied for real-time services and common channel transmission for non real-time services with varying processing gain. The effect of model parameters and user characteristics such as packet length distribution on the system behavior is quantified.

The analysis and observations presented in this thesis create a variety of research and development problems, e.g., automated performance management in order to provide adaptive network/service control, for resource allocation planning and fault management purposes in the challenging environment of wireless networks.

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